Dialogic®

Dialogic® 1000 and 2000 Media Gateway Series

User's Guide

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Revision History

This revision history summarizes the changes made in each published version of this document.

Document No.	Publication Date	Description of Revisions
64-0346-12	November 2012	Updated to support Version 6.0 SU9 Software. Chapter 1, "Overview" added new DMG2060DTISQV34 model listed under DMG2000 section in Product Description. Chapter 3, "Parameter Reference" added new parameter in Audio Group for Unhold on RE-INVITE with No SDP. Chapter 3, "Parameter Reference" added new parameters in Fax Group for Enable V.34 Fax and V.34 Fax Speed Limit. Chapter 3, "Parameter Reference" added new parameter in T.38 Fax Advanced Settings for Fax TDM Drop Out Bridge Time (DMG2000 Only). Chapter 3, "Parameter Reference" removed obsoleted parameter in T.38 Fax Advanced Settings for Maximum UDPTL Packet Size. Chapter 3, "Parameter Reference" added new parameter in Non-Menu (Hidden) Parameters for Port Disable Timeout.
64-0346-11	September 2011	Updated to support Version 6.0 SU8 Software. Chapter 3, "Parameter Reference" added new Serial Ports Group section with new parameter for Maintenance Port Enabled. Chapter 3, "Parameter Reference" added new parameter in User-Agent Group for User-Agent Header Value. Chapter 3, "Parameter Reference" added new User-User Information Group section with new parameters for User-User Enabled and Translation Method. Chapter 3, "Parameter Reference" added new ISUP Using Multipart MIME Group section with new parameter for France Telecom's IE77. Chapter 3, "Parameter Reference" updated parameter Allowed Values for Telephony Port Interface Side in T1/E1 Mode Group. Chapter 3, "Parameter Reference" added new Rx/Tx Gain Control Group section with new parameter for Analog Receive Gain. Chapter 3, "Parameter Reference" added new parameter in T1/E1 Mode Group for Clock Control. Chapter 3, "Parameter Reference" added notes to parameter for Multiple Diversion Processing in T1 ISDN Protocol Group (ISDN Signaling Mode). Chapter 3, "Parameter Reference" added notes to parameter for Multiple Diversion Processing in E1 ISDN Protocol Group (ISDN Signaling Mode). Chapter 3, "Parameter Reference" updated names of values for Optiset_300 to Telephony Switch Type in TDM Digital Parameters to be more clear between North American and European configurations. Chapter 3, "Parameter Reference" updated parameter to Default Value = -43dBm for Fax Modem Carrier Detect Threshold (DMG1000 Only) in T.38 Fax Advanced Settings. Chapter 5, "Routing Table" added more information to Select Primary / Alternate Route in Inbound TDM Rules Table and Select Primary / Alternate Route in Inbound VoIP Rules Table to show examples of failure that may occur before the call is connected to the outbound destination.

Document No.	Publication Date	Description of Revisions
64-0346-10	November 2010	Updated to support Version 6.0 SU7 Software. Chapter 3, "Parameter Reference" added new Privacy Headers Group section with new parameters for Enable Privacy Headers and Privacy Header Method. Chapter 3, "Parameter Reference" added new parameter in DSP Settings Parameters for VoIP Packet Loss Concealment (DMG2000 Only).
64-0346-09	July 2010	Updated to support Version 6.0 SU6 Software. Chapter 3, "Parameter Reference" updated parameter description for Call as Domain Name? in User-Agent Group. Chapter 3, "Parameter Reference" updated parameter description for Proactive DNS Monitoring in Monitoring Group. Chapter 3, "Parameter Reference" added new Early Media Group section in VoIP Media Parameters. Chapter 3, "Parameter Reference" added new parameters in Early Media Group for Send Early 183 Progress Response and Early Media Requires Reliable Provisional Responses. Chapter 3, "Parameter Reference" added new parameters in Audio Group for Continue Ringback on CN, Packet Time for Inbound VoIP, Telephone-Event Payload Type, and G722 Enable. Chapter 3, "Parameter Reference" added new value RFC2833-LowLatency to Digit Relay Mode in Audio Group. Chapter 3, "Parameter Reference" added new parameters in Fax Group for Fax Server Address and Fax Server Network Group. Chapter 3, "Parameter Reference" added new TDM Cause Codes Group section to support TDM Cause Codes feature. Chapter 3, "Parameter Reference" added new parameter in Non-Menu (Hidden) Parameters for DNS NAPTR and SRV Record Enable. Chapter 5, "Routing Table" added note to VoIP Host Groups Table that the maximum number of supported VoIP Host Groups is 120 (DMG2000 only).
64-0346-08	February 2010	Updated to support Version 6.0 SU5 Software. Chapter 3, "Parameter Reference" updated parameter description for VoIP Host Monitor Interval in Monitoring Group. Chapter 3, "Parameter Reference" added new parameter in Audio Group for Acceptable Media. Chapter 3, "Parameter Reference" moved parameters Unauthenticated SRTP Enable, UnEncrypted SRTCP Enable, and UnEncrypted SRTP Enable previously listed in Non-Menu (Hidden) Parameters to SRTP Group. Chapter 3, "Parameter Reference" added TDM Passthrough selection to ISDN Call Type Rules. Chapter 3, "Parameter Reference" added new parameter in DSP Settings Parameters for TDM to TDM Media Always Clear Mode (DMG2000 Only). Chapter 3, "Parameter Reference" added new parameter in Non-Menu (Hidden) Parameters for Incompatible Message STATUS. Chapter 3, "Parameter Reference" added new parameter in Non-Menu (Hidden) Parameters for ISDN Service Class. Chapter 5, "Routing Table" updated screen shots and descriptions in Inbound TDM Rules Table and Inbound VoIP Rules Table to show support for Move Selected Row buttons: Up, Down, or To Position.
64-0346-07	November 2009	Updated to support Version 6.0 SU3.2 Software.

Document No.	Publication Date	Description of Revisions
64-0346-06	July 2009	Updated to support Version 6.0 SU4 Software. Chapter 3, "Parameter Reference" updated parameter to Default Value = 90 for TCP Inactivity Timer in TCP/UDP Group. Chapter 3, "Parameter Reference" added new VoIP Network Group Parameters to support VoIP Network Group feature. Chapter 3, "Parameter Reference" added new VoIP Authentication Parameters to support VoIP Authentication feature. Chapter 3, "Parameter Reference" added hidden parameters for Routing Proxy Group Name and Routing Table Only On Proxy Down in Non-Menu (Hidden) Parameters. Chapter 5, "Routing Table" added Call Type Property Matching section to Inbound TDM Request Matching. Chapter 5, "Routing Table" added Route Method selection to Outbound Routes in Inbound VoIP Rules Table. Chapter 5, "Routing Table" added Network Group selection to VoIP Host Groups Table.
64-0346-05	May 2009	Updated to support Version 6.0 SU3.1 Software. Chapter 3, "Parameter Reference" added new parameter in Timing Group for T1 Multiplier. Chapter 3, "Parameter Reference" added new value Nortel_DMS-100 to ISDN Protocol Variant in T1 ISDN Protocol Group (ISDN Signaling Mode). Chapter 3, "Parameter Reference" added note to Enable Failover that this parameter is only applicable for DMG2060DTISQ and DMG2120DTISQ models in T1 ISDN Protocol Group (ISDN Signaling Mode). Chapter 3, "Parameter Reference" added note to Enable Failover that this parameter is only applicable for DMG2060DTISQ and DMG2120DTISQ models in E1 ISDN Protocol Group (ISDN Signaling Mode).
64-0346-04	March 2009	Updated to support Version 6.0 SU3 Software. Chapter 3, "Parameter Reference" added new parameter in User-Agent Group for Reliable Provisional Responses. Chapter 3, "Parameter Reference" added new parameter in TLS Group for Verify TLS Peer Certificate Purpose. Chapter 3, "Parameter Reference" added new parameter in T1 CAS Protocol Group (T1 CAS Signaling Mode) for Inband Type I CID to First Ring Timeout. Chapter 3, "Parameter Reference" added new parameter in CPID Settings Group for CID to First Ring Timeout. Chapter 3, "Parameter Reference" added new parameter in T1 ISDN Protocol Group (ISDN Signaling Mode) for Multiple Diversion Processing. Chapter 3, "Parameter Reference" added new parameter in E1 ISDN Protocol Group (ISDN Signaling Mode) for Multiple Diversion Processing. Chapter 3, "Parameter Reference" added new parameters in Non-Menu (Hidden) Parameters for Local Phone Context, Remote Phone Context, and User Phone Parameter. Chapter 8, "Unit Status" removed details from MIB-II Status Information section and replaced with link to SNMP Application Note.

Document No.	Publication Date	Description of Revisions
64-0346-03	January 2009	Updated to support Version 6.0 SU2 Software. Chapter 3, "Parameter Reference" added new parameter in User-Agent Group for Reliable Provisional Responses. Chapter 3, "Parameter Reference" updated parameter to Default Value = Off for Signaling Digit Relay Mode in Audio Group. Chapter 3, "Parameter Reference" updated parameter to remove the statement "Early Media is supported for VoIP to TDM calls only" since it is now supported in both directions for RFC 3960 Early Media Support in Audio Group. Chapter 3, "Parameter Reference" added new parameter in TDM General Parameters for Connect Outbound Call On DTMF. Chapter 3, "Parameter Reference" updated parameter to include more supported Allowed Values for Network Specific Facilities (NSF) in T1 ISDN Protocol Group (ISDN Signaling Mode). Chapter 3, "Parameter Reference" added new TDM Call Type Group section to support call type per call feature.
64-0346-02	September 2008	Updated to support Version 6.0 SU1 Software. Chapter 3, "Parameter Reference" updated parameter description for Signaling Digit Relay Mode in Audio Group. Chapter 3, "Parameter Reference" added new parameter in T1 CAS Protocol Group (T1 CAS Signaling Mode) for Transfer Feature Code. Chapter 3, "Parameter Reference" added new parameter in Feature Code Group for Transfer Feature Code.

Document No.	Publication Date	Description of Revisions
64-0346-01	March 2008	Updated to support Version 6.0 Software. Global Updates: Added and revised screen shots for the enhanced Web interface. Added new DMG2060DTISQ and DMG2120DTISQ models which include support for survivability. Removed some parameters that are obsoleted in Version 6.0 Software. Chapter 3, "Parameter Reference" added new parameter in Server Group for DNS Server Address 2. Chapter 3, "Parameter Reference" added new parameter in Monitoring Group for VoIP Host Monitor Interval. Chapter 3, "Parameter Reference" added new parameter in T1/E1 Mode Group for Telephony Port Interface Side. Chapter 3, "Parameter Reference" added new parameter in T1 CAS Protocol Group (T1 CAS Signaling Mode) for Enable Glare Detection. Chapter 3, "Parameter Reference" added new parameters in T1 ISDN Protocol Group (ISDN Signaling Mode) for ISDN Answer Supervision Enable, Network Specific Facilities (NSF), and Enable Failover. Chapter 3, "Parameter Reference" added new parameters in E1 ISDN Protocol Group (ISDN Signaling Mode) for ISDN Answer Supervision Enable and Enable Failover. Chapter 3, "Parameter Reference" added new parameter in TDM General Parameters for Disconnect on Fax Cleardown Tone. Chapter 3, "Parameter Reference" added new parameter in T.38 Fax Advanced Settings for Fax Modem Carrier Detect Threshold (DMG1000 Only). Chapter 3, "Parameter Reference" added new section and parameters in Tone Generation Configuration Parameters. Chapter 3, "Parameter Reference" added new section and parameters in Positive Answer Machine Detection. Chapter 5, "Routing Table" updated with new functionality and configurations (previously referred to as the Dial Plan). Chapter 5, "Routing Table" added new section for Call Routing Examples.

About This Publication

The following topics provide information about this guide:

- Purpose
- Intended Audience
- How to Use This Publication
- Related Information

Purpose

This document provides information about installing, configuring, operating, and maintaining the Dialogic® Media Gateway.

Intended Audience

This information is intended for:

- Distributors
- System Integrators
- Value Added Resellers (VARs)
- Original Equipment Manufacturers (OEMs)

How to Use This Publication

This information is organized as follows:

- Chapter 1, "Overview" provides a description of the product and discusses call routing, address translation, and the Web interface.
- Chapter 2, "Media Gateway Configuration" provides procedures for configuring and upgrading the Media Gateway.
- Chapter 3, "Parameter Reference" lists the Media Gateway parameters that can be configured from the Web interface.
- Chapter 4, "Call Progress Tones" describes how to view, edit, learn, and validate call progress tones from the Web interface.
- Chapter 5, "Routing Table" describes a set of rules used to define the characteristics of a call routed through the Media Gateway.

- Chapter 6, "Media Gateway Parsers" describes the Media Gateway in-band Type I (on-hook) and Type II (off-hook) integration parsers for analog units, and the display parser for digital units. These parsers allow the user to define the meaning of either the in-band/on-hook integration strings or display strings received from the telephony network.
- Chapter 7, "Data Security" provides information about configuring security on the Media Gateway for HTTP, call control, and voice.
- Chapter 8, "Unit Status" describes the various types of status information that may be obtained about the Media Gateway.
- Chapter 9, "Diagnostics" provides information about using diagnostic logging, running the diagnostic tests, and using the various terminal commands to perform diagnostics on the Media Gateway.

Note: The products previously known as Intel NetStructure PBX-Media Gateway and T1/E1-Media Gateway are now Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000). For more product name changes, refer to New Product Naming Conventions.

New Product Naming Conventions

Previous Name	New Name
PBX-IP Media Gateway	Dialogic® 1000 Media Gateway (DMG1000)
PIMG	DMG1000
PIMG40LS	DMG1004LSW
PIMG80LS	DMG1008LSW
PIMG80DNI	DMG1008DNIW
PIMG80MTLDNI	DMG1008MTLDNIW
PIMG80RLMDNI	DMG1008RLMDNIW
T1/E1-IP Media Gateway	Dialogic® 2000 Media Gateway (DMG2000)
TIMG	DMG2000
TIMG300DTI	DMG2030DTIQ
TIMG600DTI	DMG2060DTIQ
TIMG1200DTI	DMG2120DTIQ
	DMG2060DTISQ (with survivability)
	DMG2120DTISQ (with survivability)

Related Information

For additional information related to the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) products, see the following:

- *Dialogic® 1000 and 2000 Media Gateway Series Getting Started Guide* for information about installing, cabling, and initializing the product prior to performing configuration and operation tasks.
- *Dialogic® Media Gateway Installation and Configuration Integration Notes* for details on typical installation and configuration of Media Gateway when used to interface between PBX and unified messaging application.
- *http://www.dialogic.com/manuals* (for Dialogic® product documentation)
- *http://www.dialogic.com/support* (for Dialogic technical support)
- *http://www.dialogic.com* (for Dialogic® product information)

Overview 1

The following information provides an overview of the Dialogic® Media Gateway:

- Product Description 25
- Call Routing in Phone Emulating Mode 30
- Voice over IP Address Translation 31
- Security 32
- Web Interface. 32
- Online Help 35

1.1 Product Description

The Media Gateway is a telephony gateway appliance that connects to phone lines through its telephony interface and connects to a LAN via a 10 BaseT or 100 BaseT Ethernet connector.

NOTE: Throughout this document, the term **Media Gateway** addresses information that applies to both the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) products. The term **DMG1000** applies only to information relating to the DMG1000 product and the term **DMG2000** applies only to information relating to the DMG2000 product.

The Media Gateway provides an inexpensive bridge between a legacy PBX or public switched telephone network (PSTN) and a managed packet network. This device converts signals from circuit switched equipment into Session Initiation Protocol (SIP) standard protocol for transmission over a local area network (LAN) or wide area network (WAN) to communications devices such as IP phones, wireless phones, and IP servers in almost any location.

The DMG1000 is available in the following models:

- **DMG1008LSW, DMG1004LSW** - Supports phone emulation mode for analog interfaces.
- **DMG1008DNIW** - Supports phone emulation mode for a number of digital PBXs, including Avaya, Nortel, NEC, and Siemens.
- **DMG1008MTLDNIW** - Supports phone emulation mode for Mitel digital PBXs.
- **DMG1008RLMDNIW** - Supports phone emulation mode for Rolm 8000 and 9751 switches.

The DMG2000 is available in the following models:

- **DMG2030DTIQ** - Supports phone emulation mode for a single T1 or E1 interface.
- **DMG2060DTIQ** - Supports phone emulation mode for two T1 or E1 interfaces.
- **DMG2120DTI**Q - Supports phone emulation mode for four T1 or E1 interfaces.
- **DMG2060DTISQ** - Supports phone emulation mode with survivability for two T1 or E1 interfaces.
- **DMG2060DTISQV34** - Supports phone emulation mode with survivability for two T1 or E1 interfaces.
- **DMG2120DTISQ** - Supports phone emulation mode with survivability for four T1 or E1 interfaces.

Note: The Item Market Name on the Media Gateway may vary slightly, depending on the version.

Depending on the model, the Media Gateway can be configured for the following operating mode:

- Phone Emulating

1.1.1 Phone Emulating

In the Phone Emulating mode, the Media Gateway operates as a telephony gateway appliance that emulates the following for transporting PBX functionality over a packet-switched network:

- up to eight station sets (DMG1000 models)
- up to 24 station sets (single T1 DMG2000 model)
- up to 48 station sets (dual T1 DMG2000 model)
- up to 96 station sets (quad T1 DMG2000 model)
- up to 30 station sets (single E1 DMG2000 model)
- up to 60 station sets (dual E1 DMG2000 model)
- up to 120 station sets (quad E1 DMG2000 model)

The Media Gateway translates protocols for call setup and release between the IP network and the PBX or PSTN, and converts the media formats between the two networks.

Figure 1 shows how the Media Gateway provides a gateway between voice over IP (VoIP) devices (SIP) on a LAN and the PBX. By emulating station sets to the proprietary PBX, the Media Gateway provides full call party information to the IP network.

The DMG2000 or the analog version (DMG1008LSW, DMG1004LSW) of the DMG1000 can also connect directly to the PSTN, as shown in Figure 2.

Figure 1. Typical IP Gateway Phone Emulating Topology - PBX Connection

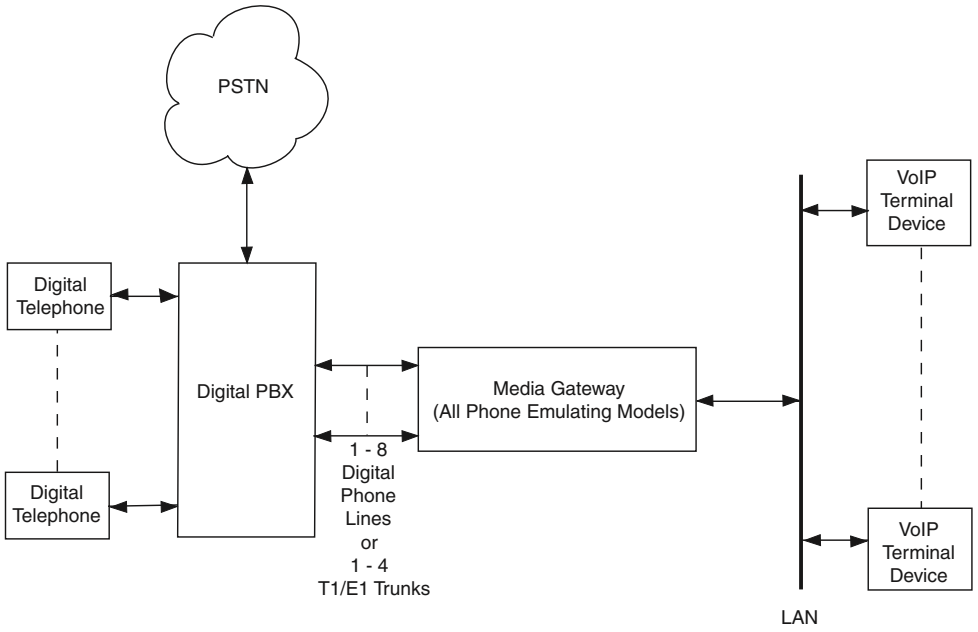
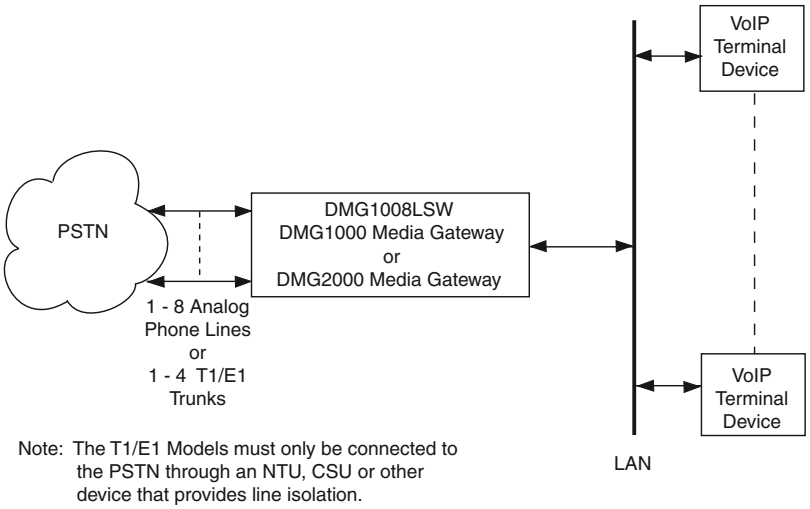


Figure 2. Typical IP Gateway Phone Emulating Topology - PSTN Connection


PSTN
DMG1008LSW DMG1000 Media Gateway or DMG2000 Media Gateway
1 - 8 Analog Phone Lines or 1 - 4 T1/E1 Trunks
VoIP Terminal Device
VoIP Terminal Device
LAN
Note: The T1/E1 Models must only be connected to the PSTN through an NTU, CSU or other device that provides line isolation.

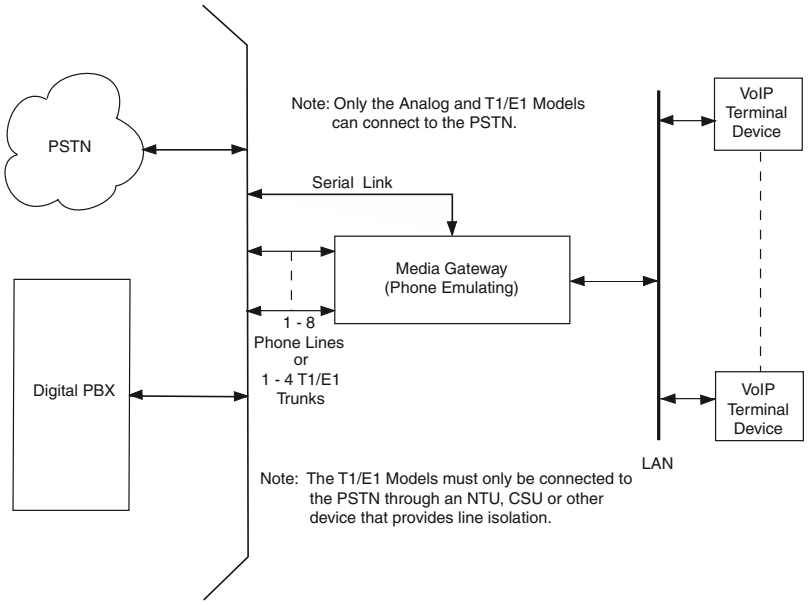
1.1.1.1 Serial Protocol Support in Phone Emulating Mode

By emulating telephone sets to the switch, the Media Gateway provides call party information over the IP network. However, the amount of call party information that the Media Gateway can provide is limited to the amount of data that the switch provides its station sets. Some proprietary switches

provide full call party information across the station set interfaces while others provide little or no call party information across the station set interface. The PBX switches that provide little or no call party information typically will provide full call party information across a separate serial interface connection.

For this reason, the Media Gateway supports a serial link interface to the switch or PSTN, and supports several serial protocols. Figure 3 shows how the Media Gateway connects to a switch or PSTN that uses a serial link to provide call party information. Using this serial link, the Media Gateway is able to provide full call party information on a PBX switch or PSTN that provides little or no call party information via their station set interfaces.

Figure 3. IP Gateway Using Serial Link


PSTN
Note: Only the Analog and T1/E1 Models can connect to the PSTN.
VoIP Terminal Device
Serial Link
Media Gateway (Phone Emulating)
1 - 8 Phone Lines or 1 - 4 T1/E1 Trunks
Digital PBX
VoIP Terminal Device
LAN
Note: The T1/E1 Models must only be connected to the PSTN through an NTU, CSU or other device that provides line isolation.

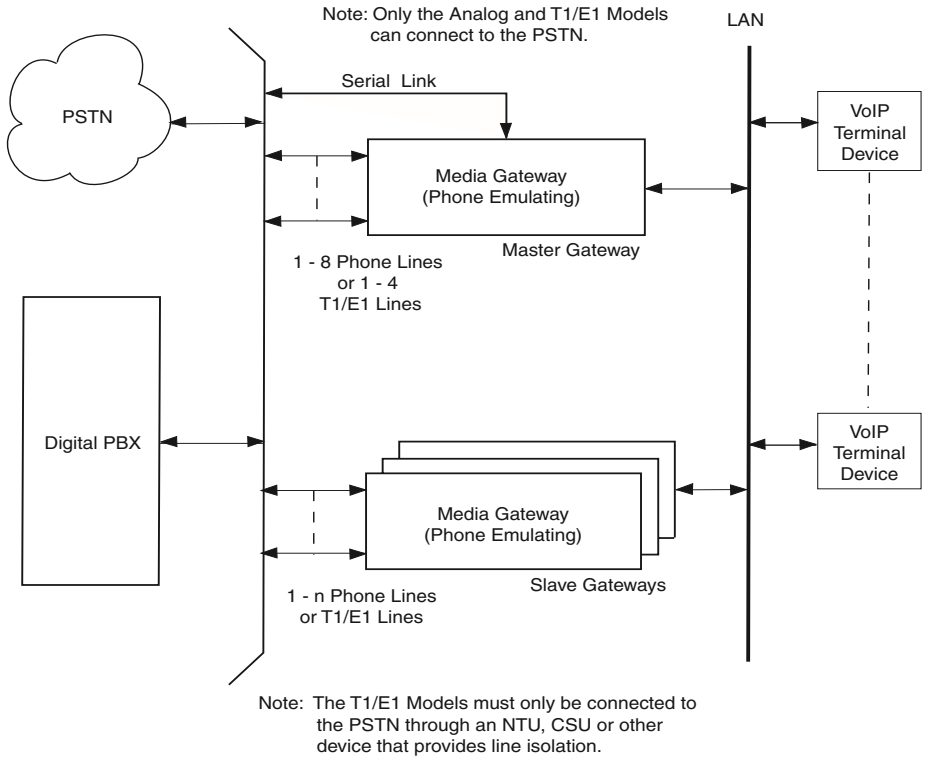
When a call arrives at a telephony port on the Media Gateway, the switch or PSTN will send a data packet across the serial link containing the call party information associated with the call. If configured to use the serial interface, the Media Gateway will use the data in the serial packet as the call party information when the call notification is sent across the IP network to the VoIP terminal device that the call is intended for - superseding any call information that may have arrived across the station set interface. Finally, the serial protocols also support the message waiting indication (MWI) feature that allows the Media Gateway to control message indications on telephone sets connected to the switch or PSTN.

For DMG1000 types, the serial link from the switch is connected to the serial port interface on the DMG1000 (DIAGNOSTICS connector). Table 1, “DIAGNOSTICS Connector Pin Designations” in the *Getting Started Guide* shows the connector pin designations.

For DMG2000 types, the serial link from the switch is connected to the serial port interface on the DMG2000 (COM 1 or COM 2 connector). Table 4, “COM 1 and COM 2 Connector Pin Designations” in the *Getting Started Guide* shows the connector pin designations.

The switch only provides a single serial link connection. At a site where there are multiple Media Gateway units, only one of the units can be physically connected to the switch or PSTN serial link. This unit is the serial protocol Master Media Gateway while the remaining units are considered serial protocol slaves. Figure 4 shows how multiple Media Gateways connect to a switch or PSTN that uses a serial link to provide call party information. It is the responsibility of the Master gateway to send all serial link data intended for Slave Gateways to the Slave Gateways across the IP network. Similarly, anytime a Slave Gateways needs to communicate to the switch or PSTN across the serial link, the slave unit sends the data across the IP link to the Master Gateway. The Master Gateway will then send the data across the serial link on behalf of the slave device.

Figure 4. Multiple IP Gateways Using Serial Link


Note: Only the Analog and T1/E1 Models can connect to the PSTN.
LAN
Serial Link
PSTN
VoIP Terminal Device
Media Gateway (Phone Emulating)
Master Gateway
1 - 8 Phone Lines or 1 - 4 T1/E1 Lines
Digital PBX
VoIP Terminal Device
Media Gateway (Phone Emulating)
Slave Gateways
1 - n Phone Lines or T1/E1 Lines
Note: The T1/E1 Models must only be connected to the PSTN through an NTU, CSU or other device that provides line isolation.

The Media Gateway supports the following serial protocols:

- SMDI (Simple Message Desk Interface)
- MCI (NEC Systems only)
- MD110 (Ericsson Systems only)

Both the serial interface and the selection of which serial protocol to use are configurable using the Web interface. Refer to Chapter 3, "Parameter Reference" for information about configuring the serial interface and serial protocols.

1.2 Call Routing in Phone Emulating Mode

The Media Gateway routes calls from the Switch network to a VoIP destination on the IP network. Conversely, it routes calls from the IP network through a Switch port to a destination telephone number on the Switch network. The Media Gateway supports the following call routing options:

- User configurable list of VoIP Servers
- IP load Balancing
- IP Fault Tolerance

In its simplest form, call routing is supported by configuring a single VoIP Server to receive and/or originate calls through the Media Gateway. In this manner, all inbound Switch-to-IP calls will be sent to the single user configured VoIP endpoint. For IP-to-Switch calls, the telephony port will be selected in a round robin fashion – where each IP-to-Switch call will be routed to the next available telephony port.

If the user configures more than one VoIP Server to receive and/or originate calls through the Media Gateway, then the user has the option to have the incoming Switch-to-IP calls load balanced between the configured VoIP Servers. Specifically, incoming Switch calls will be routed to a VoIP server in a round-robin fashion. For example, if there are three (3) VoIP Servers configured, the first call will be routed to the first VoIP server, the second call will be routed to the second VoIP server, the third call will be routed to the third VoIP server. The next call will be routed to the first VoIP Server and the process will start all over again. IP-to-Switch calls are handled in the same way when multiple VoIP Servers are configured as when only a single VoIP server is configured (e.g. in a round robin fashion).

If the user configures more than one VoIP Server to receive and/or originate calls through the Media Gateway, then the user has the option to support fault tolerance on the incoming Switch-to-IP calls. Specifically, if the VoIP server fails to respond to incoming Switch call (or responds with an error), the Media Gateway will route the call to the next VoIP Server in the user configurable list of VoIP Servers.

Note: If both IP Load Balancing and Fault Tolerance are enabled, then incoming IP-to-Switch calls will be routed to the configured VoIP Servers in a round robin fashion and, if at any time a VoIP Server fails to respond or responds with any error, the Media Gateway will route the call to the next available VoIP Server.

1.2.1 Un-Routable Calls

A call is un-routable by the Media Gateway if the unit is unable to route the call to the other network. This may occur if there is insufficient destination address information to determine a destination for the call, or if there are not enough free resources on the Media Gateway to route the call. The Media Gateway provides a number of different ways to handle these conditions in order to insure that no calls are dropped or not completed.

1.2.1.1 IP to Switch Calls

There are a few circumstances where an inbound IP call may not be routed to the Switch network. They are the following:

- No available Switch Ports
  - There are no available Media Gateway Switch ports on which to carry the call.
- Invalid or No Switch destination address specified
  - The IP call information does not contain a valid E.164 destination address for the Switch network.

The administrator may specify a default destination IP address that is to receive any inbound IP calls that cannot be routed to the Switch. If the administrator specifies a default destination IP address for un-routable IP calls, then any inbound IP call that cannot be routed to the Switch network is handled in the Routing Table as alternate routes.

1.2.1.2 Switch to IP Calls

There are a few circumstances in which an inbound Switch call may not be routed to the IP network. They are the following:

- IP destination not configured.
- IP destination not present.

The administrator may specify a default destination Switch extension that is to receive any inbound Switch calls that cannot be routed to the IP network. If the administrator specifies a default destination Switch extension for un-routable Switch calls, then any inbound Switch call that cannot be routed to the IP network is handled in the Routing Table as alternate routes.

1.3 Voice over IP Address Translation

The Voice over IP (VoIP) Address Translator provides network services to SIP devices such as the Media Gateway. SIP devices register with the VoIP Address Translator to sent and receive SIP calls.

The VoIP Address Translator can provide network services such as:

- Controlling the number and type of connections allowed across the network.
- Helping to route a call to the correct destination.
- Determining and maintaining the network address for incoming calls.

Without a VoIP Address Translator, all IP destination addresses must be specified to the Media Gateway as IP v4 addresses (e.g. 10.10.4.128). IP terminal devices must also explicitly specify the IP address of the Media Gateway as the desired gateway when originating PBX calls.

With a VoIP Address Translator, IP destination addresses may be specified to the Media Gateway as e-mail addresses, alphanumeric aliases, E.164 telephone numbers, domain names, and any other format supported by the VoIP Address Translator. IP terminal devices may explicitly specify the IP

address of the Media Gateway as the desired gateway, or they may rely on the VoIP Address Translator to address the Media Gateway using the gateway prefix configured on the Media Gateway.

1.4 Security

The Media Gateway supports two kinds of security:

- Access Security
- Data Security

1.4.1 Access Security

Access to the Media Gateway is secured by requiring a user name and password to login to the gateway. The user name and password are required for all interfaces of the Media Gateway (Web interface, serial interface, and telnet interface).

1.4.2 Data Security

Data security for the Media Gateway includes the use of various secure protocols when transmitting and receiving data. The Media Gateway supports security for three types of data:

- HTTP security - Data transmitted between the Media Gateway and a Web browser. To secure HTTP, the Media Gateway uses HTTPS protocol.
- Call Control security - Data used to setup and tear down a call. To secure Call Control, the Media Gateway uses Transport Layer Security (TLS) on top of SIP.
- Voice security - The actual conversation once a call is connected. To secure voice, the Media Gateway uses SRTP.

Because, for security, the HTTPS and TLS protocols also require digital identity certificates (e.g. public key certificates), Certificate Configuration and Management is also provided by the Media Gateway.

For additional information about how the Media Gateway supports security, see Chapter 7, “Data Security”.

1.5 Web Interface

The Web interface is accessed from a workstation on the Ethernet connected to the **LAN** connector on the rear panel of the Media Gateway. Communication is established by starting a Web browser at the workstation and entering the IP address of the Media Gateway. The unit's Web interface is password protected and the password can be changed by the system administrator. Figure 5 shows a typical Web interface page for a DMG1000 unit and Figure 6 shows a typical Web interface page for a DMG2000 unit.

Figure 5. DMG1000 Web Interface

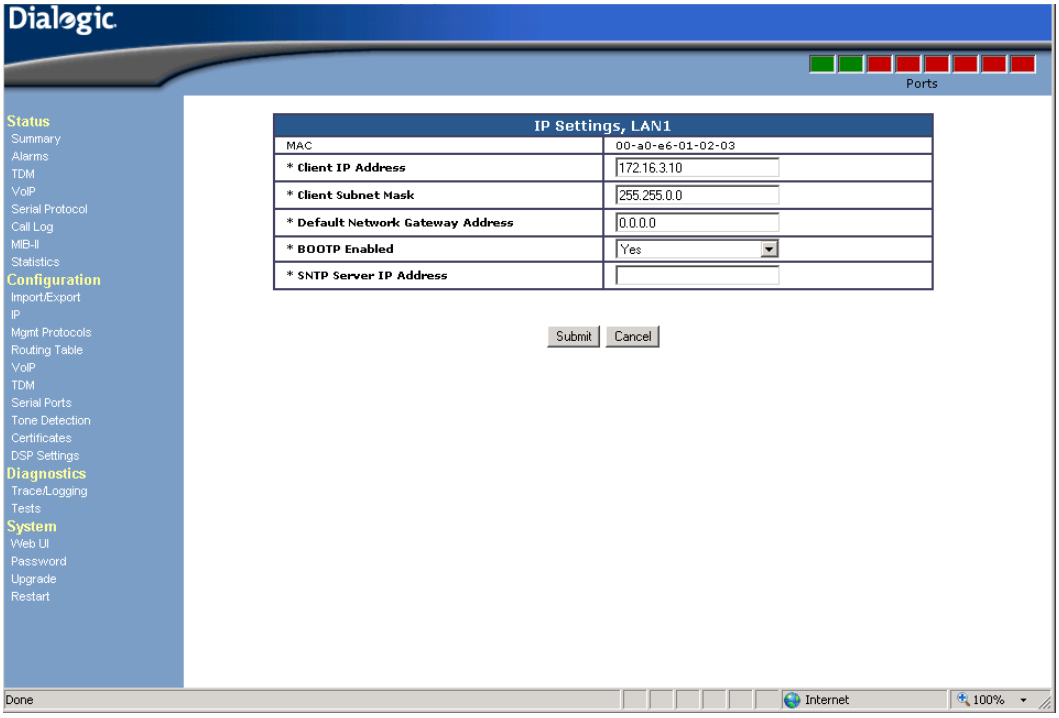
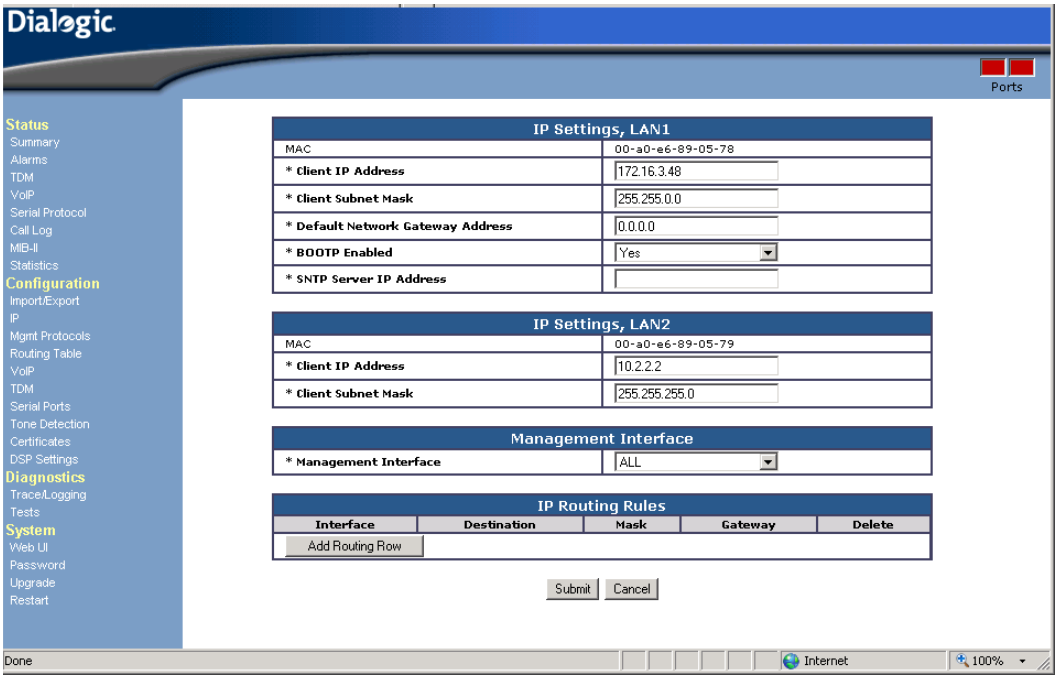


Figure 6. DMG2000 Web Interface



The Media Gateway's Web interface is divided into four sections, **Status**, **Configuration**, **Diagnostics**, and **System**.

The **Status** Web pages provide run-time information and statistics about the operation of the unit.

The **Diagnostics** Web pages provide the system administrator with an interface to perform diagnostics tasks.

The **Configuration** Web pages provide the system administrator with an interface to configure the Media Gateway. The Configuration menu selections include:

- **Import/Export** - Selecting Import/Export brings up the Import/Export Web page which allows you to save a copy of your current configuration (Export) to a directory on the computer or to import a configuration file to the Media Gateway (Import). For information about exporting a configuration file, see Section 2.6.1, "Exporting Configuration Information", on page 41. For information about importing a configuration file, see Section 2.6.2, "Importing Configuration Information", on page 42.
- **IP** - Selecting IP brings up the IP Web page which allows you to configure the IP parameters. For detailed information about the IP parameters, see Section 3.1, "IP Settings", on page 46.
- **Management Protocols** - Selecting Management Protocols brings up the Management Protocols Web page which allows you to configure E-mail, Syslog, SNMP, Web Server, and Telnet parameters. For detailed information about the Management Protocols parameters, see Section 3.2, "Management Protocols Parameters", on page 52.
- **Routing Table** - Selecting Routing Table brings up the Routing Table Web page which allows you to configure a set of rules used to define the characteristics of a call routed through the gateway. For detailed information about the Routing Table, see Section 5.2, "Router Configuration", on page 195.
- **VoIP** - Selecting VoIP brings up the VoIP Web page which allows you to configure VoIP parameters. For detailed information about the VoIP parameters, see Section 3.3, "VoIP General Parameters", on page 59.
- **TDM** - Selecting TDM brings up the TDM Web page which allows you to configure TDM parameters. Some of these parameters differ, depending on the different DMG1000 and DMG2000 models. For detailed information about the TDM parameters, see Section 3.7, "TDM General Parameters", on page 105.
- **Serial Ports** - Selecting Serial Ports brings up the Serial Ports Web page. This Web page allows you to configure the serial protocol for the Media Gateway serial port. See Section 3.15, "Serial Ports Parameters", on page 146 for detailed information about the Serial Ports parameters.
- **Tone Detection** - Selecting Tones allows you to manipulate call progress tone parameters. These parameters define the characteristics (frequencies, durations, and deviations) of the tones that the Media Gateway detects during call progress analysis. For more information, see Section 3.17, "Tone Detection Parameters", on page 154 and Chapter 4, "Call Progress Tones".
- **Certificates** - Selecting Certificates brings up the Certificates Web page which allows you to configure Certificates parameters. For detailed information about the Certificates parameters, see Section 3.18, "Certificates Parameters", on page 161.

- **DSP Settings** - Selecting DSP Settings brings up the DSP Settings Web page which allows you to configure DSP Settings parameters. For detailed information about the DSP Settings parameters, see Section 3.19, "DSP Settings Parameters", on page 162.

The **System** Web pages provide the system administrator with an interface to change password, upgrade system, and restart system.

- **Password** - Selecting Password brings up the Password Web page. This page allows you to change your password. For information about the procedure for changing your password, see Section 2.3, "Changing the Password", on page 39.
- **Upgrade** - Selecting Upgrade brings up the Upgrade Web page. The Upgrade Web page allows you to upgrade the Media Gateway software. For information about the procedure for upgrading the software, see Section 2.7, "Upgrading the Software", on page 42.
- **Restart** - Selecting Restart from the Configuration menu brings up the Restart Web page which allows you to restart the Media Gateway. Restarting the unit is required when certain parameter values are changed. You have two options to choose from when restarting the unit:
  - Restart Unit Now - Clicking this button will cause the unit to restart immediately.
  - Restart Unit When Idle - Clicking this button will cause the unit to restart when the unit is considered in the idle state. In the idle state, there are no calls (incoming, outgoing, or connected) on any of the PBX ports of the Media Gateway. By selecting this option, you will schedule a restart time that minimizes the effect the restart will have on call traffic through the unit.

1.6 Online Help

Context-sensitive online Help is provided for the various Web pages. When you move the mouse pointer over a parameter name, the pointer changes from an arrow to a "?" symbol and Help information about that parameter is displayed at the top of the page.

Media Gateway Configuration 2

Information about configuring, and upgrading the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) is contained in the following sections:

- Setting the IP Address 37
- Basic Configuration Via the Serial Port 38
- Changing the Password 39
- Configuration Procedure 39
- Restart Options 40
- Importing and Exporting Configuration Information 40
- Upgrading the Software 42

Note: Refer to the *Getting Started Guide* for information about initially logging on to the Media Gateway and performing basic configuration via the serial port.

2.1 Setting the IP Address

Once the IP address of the Media Gateway unit has been configured through the Web interface, the unit must be restarted to activate the new configuration. The Media Gateway unit will then be accessible to the Web browser at the newly configured IP address. All configuration parameters are saved in the Media Gateway's non-volatile memory.

Use the following steps to assign the Media Gateway a unique IP address:

1. Select the IP configuration Web page from the **Configuration** menu on the left side of the Web page.
2. Change the unit's IP address from the default address by entering the new IP address in the **Client IP Address** box.
3. Configure the subnet mask if it is different from the default value by entering the new subnet mask in the **Client Subnet Mask** box.
4. Configure the IP address of the default network gateway router by entering the IP address in the **Default Network Gateway Address** box.
5. Click on the **Apply Changes** button to save the configuration in the database.

Note: Refer to Chapter 3, "Parameter Reference" for a description of all of the configuration parameters that may be changed.

6. For the configuration change to take effect, you will be prompted to restart the Media Gateway by clicking on Restart on the Web page or by selecting **Restart** from the **Configuration** menu.
7. When the **Restart** Web page appears, click on **Restart Unit Now** to restart the Media Gateway.

8. Once the system completes its initialization (after approximately one minute), browse to the new IP address. The Web browser can now access the unit at the new IP address.
9. If it was necessary to change the IP address of your Windows® workstation in the Initial Log On procedure (described in the *Getting Started Guide*), you should now change it back to the original IP address and access the Media Gateway using the Media Gateway's newly configured IP address.
10. You must now log on to the system again. After logging on, the Summary Web page will now appear. You may now select any item from the **Status** or **Configuration** menu.

2.2 Basic Configuration Via the Serial Port

When the default IP address is unreachable (or if a previously configured IP address becomes unreachable) on your network, use the following procedure to set the Client IP Address, Client Subnet Mask, Default Network Gateway Address, Operating Mode, and PBX Type parameters of the Media Gateway:

1. Connect a serial cable to the serial connector on the rear panel of the Media Gateway unit (DIAGNOSTICS connector on the DMG1000 models or the COM 2 connector on the DMG2000 models). For connector pin designation information, refer to the DIAGNOSTICS Connector Pin Designations table or the COM 1 and COM 2 Connector Pin Designations table in the *Getting Started Guide*.
2. Using a standard serial interface application (for example, Procomm Plus or HyperTerminal), set the workstation to the following:
   - Baud Rate = 38400 (DMG1000) or 115200 (DMG2000) bps
   - Parity = None
   - Data Bits = 8
   - Stop Bits = 1
   - Hardware Flow Control = Off
3. Press the Enter key repeatedly until the following prompt appears:

   PIMG>
4. At the prompt, type **pwd** and press Enter.
5. When prompted, enter the password for the admin user (the default is **IpodAdmin**) and press Enter.
6. At the prompt, type **quickcfg** and press Enter.
7. You will then be prompted to enter the following Media Gateway parameter information:
   - Client IP Address (See Client IP Address parameter information in Section 3.1, "IP Settings", on page 46.)
   - Client Subnet Mask (See Client Subnet Mask parameter information in Section 3.1, "IP Settings", on page 46.)
   - Network Gateway IP (if required) (See Default Network Gateway Address parameter information in Section 3.1, "IP Settings", on page 46.)

- For DMG1000 models, select Telephony Switch Type (See Telephony Switch Type parameter information in Section 3.10, "TDM Digital Parameters", on page 135.)

Note: The Telephony Switch Type parameter does not apply to the Models DMG1008MTLDNIW, DMG1008LSW, DMG1004LSW, DMG2030DTIQ, DMG2060DTIQ, and DMG2120DTIQ.

- For DMG2000 models, select:
  - Line Mode (Section 3.8.1.1, "Line Mode", on page 109)
  - CAS Protocol (Section 3.8.1.2, "Signaling Mode", on page 109 and Section 3.8.2.1, "T1 CAS Protocol", on page 111) or ISDN Protocol (Section 3.8.1.2, "Signaling Mode", on page 109 and Section 3.8.3.4, "ISDN Protocol", on page 118)

8. When prompted that the parameters have been successfully configured, type **restart** at the PIMG> prompt to restart the Media Gateway.

   You should now be able to connect to the Media Gateway from the Web Interface using the newly configured IP Address.

2.3 Changing the Password

The steps for changing the password are described in the following procedure:

1. Start your Web browser.
2. In the Web browser address box, enter the IP address of the Media Gateway that you wish to access.
3. When the **System Login** Web page appears, enter the user name and current password in the boxes provided and click on the **Log On** button.

Note: The user name and password are case sensitive.

4. Once the login has been accepted, the Media Gateway Status/Configuration Web page will appear. Select the **Password** Web page from the **Configuration** menu on the left side of the page.
5. Enter the current password in the Old Password box.
6. Enter the new password in the New Password box.
7. Enter the new password a second time in the Confirm box. Then, click on the **Change** button to replace the old password with the new password.

2.4 Configuration Procedure

The Media Gateway is configured using a Web interface.

To change configuration parameter values through the Web interface, use the following procedure:

1. From a workstation connected to the Media Gateway via the Ethernet, start the Web browser.
2. Enter the IP address assigned to the Media Gateway. For example:

```
http://10.12.13.74
```

3. When the System Login Web page appears, enter **admin** in the User Name box and your password in the Password box, then click on the **Log On** button.

Note: The user name and password are case sensitive.

4. Select the appropriate configuration Web page from the **Configuration** menu. For example, to define the PCM Coding as aLaw, select the **System** Web page from the menu.

   When the **System** Web page appears, select **aLaw** from the drop down box as the PCM Coding parameter value.

5. Click on the **Apply Changes** button to save the new configuration in the database, or click on the **Reset** button to return the parameter to the previous value.

   Note: If the parameter whose value you are changing requires a restart for the change to take effect, you will be prompted by a flashing **Restart Required** blue box in the upper left-hand corner of the screen.

6. If a restart is required, click on Restart on the System screen or select **Restart** from the **Configuration** menu.
7. When the **Restart** Web page appears, click on **Restart Unit Now** or **Restart Unit When Idle** to restart the Media Gateway. (See Section 2.5, "Restart Options", on page 40).
8. Once the system completes its initialization (after approximately one minute), select **Refresh** from the Web Browser **View** Menu.
9. You must now log on to the system again. After logging on, the Summary screen will appear. You may now select any item from the **Status** or **Configuration** menu.

2.5 Restart Options

Selecting Restart from a configuration screen or selecting **Restart** from the **Configuration** menu brings up the **Restart** Web page which allows you to restart the Media Gateway. Restarting the unit is required when certain parameter values are changed. You have two options to choose from when restating the unit:

- **Restart Unit Now** - Clicking this button will cause the unit to restart immediately.
- **Restart Unit When Idle** - Clicking this button will delay the unit from restarting until the unit is considered in the idle state. In the idle state, there are no calls (incoming, outgoing, or connected) on any of the PBX ports of the Media Gateway. By selecting this option, you will schedule a restart time that minimizes the effect the restart will have on call traffic through the unit.

2.6 Importing and Exporting Configuration Information

The Import/Export Web page allows you to import or export Media Gateway configuration information to a *.ini file on the host machine connected to the Media Gateway via the Web interface.

Scenarios where importing or exporting the configuration information may be useful include:

- **Backup configuration** - It is recommended that, after configuring the Media Gateway, you backup the configuration information by exporting it to a *.ini file. If at some time in the future

the configuration has been changed and now you wish to return to the previous configuration, you can import the previous configuration data back to the Media Gateway.

- **Multiple unit implementation** - At a site where there are multiple Media Gateway units, much of the configuration will be the same between units (e.g. Audio Parameters, SIP Parameters, etc.). An installer could configure the first unit, export the *.ini file, edit the unit specific parameters in the *.ini file (e.g. IP Address) using Notepad, and then import the modified file to the next unit. This will speed up and simplify the installation procedure at larger sites.
- **Support** - If the Web interface is inaccessible to customer support, you can export the configuration information to a *.ini file and then email the file to customer support for them to view the configuration directly.
- **Software upgrades** - Normally, software upgrades will not affect the configuration. When a major software upgrade is performed, however, existing configuration information may be erased and default settings restored. In this scenario, the current configuration information could be exported before the upgrade is performed and then imported back to the Media Gateway after the upgrade. All of the previous configuration data will then be restored, except for the new parameters, which would be at their default settings.

2.6.1 Exporting Configuration Information

To export the current configuration information:

1. Select the **Import/Export** Web page from the **Configuration** menu.
2. There are options to export all settings, export subset of settings (only tone definitions, only routing table, only CPID parsing rules), or export routing table schema.
   a. Click the **Export All Settings** button in the **Export Files** box. The File Download dialog box then appears.

      You may click the **Open** button to view/edit the configuration file in Notepad or click the **Save** button to download and save the file to a directory. The default name for the file is *config.ini*.

      Clicking the **Save** button causes the Save As dialog box to appear.

   b. Click the **Only Tone Definitions** button in the **Export Files** box. The File Download dialog box then appears.

      You may click the **Open** button to view/edit the configuration file in Notepad or click the **Save** button to download and save the file to a directory. The default name for the file is *tones.ini*.

      Clicking the **Save** button causes the Save As dialog box to appear.

   c. Click the **Only Routing Table** button in the **Export Files** box. The File Download dialog box then appears.

      You may click the **Open** button to view/edit the configuration file in Notepad or click the **Save** button to download and save the file to a directory. The default name for the file is *dmg.xml*.

      Clicking the **Save** button causes the Save As dialog box to appear.

d. Click the **Only CPID Parsing Rules** button in the **Export Files** box. The File Download dialog box then appears.

   You may click the **Open** button to view/edit the configuration file in Notepad or click the **Save** button to download and save the file to a directory. The default name for the file is *cpid.adt*.

   Clicking the **Save** button causes the Save As dialog box to appear.

e. Click the **Export Routing Table Schema** button in the **Export Files** box. The File Download dialog box then appears.

   You may click the **Open** button to view/edit the configuration file in Notepad or click the **Save** button to download and save the file to a directory. The default name for the file is *dmg.xsd*.

   Clicking the **Save** button causes the Save As dialog box to appear.

3. To save the configuration file to a directory, click the **Save** button. In the Save As dialog box, choose the directory in which you wish to save the configuration file and, if you wish, rename the file. Do not, however, change the file type from INI.

2.6.2 Importing Configuration Information

To import a configuration file:

1. Select the Import/Export Web page from the **Configuration** menu.
2. Enter an INI file name in the Import box or use the **Browse...** button to select an INI file.
3. Click the **Import Settings** button to import the configuration information to the Media Gateway.

2.7 Upgrading the Software

Software upgrades for the Media Gateway products will be made available on the Dialogic support Web site as needed. Contact technical support to obtain software upgrade files. Upgrade files are uploaded to the Media Gateway using the Web interface.

Caution: When a major software upgrade is performed, existing configuration information may be erased and the default values restored. It is recommended that before upgrading the software, you should export the current configuration information using the Export utility. Refer to Section 2.6, "Importing and Exporting Configuration Information", on page 40 for information about backing up the configuration information.

When an upgrade file is made available, follow these steps to upgrade the Media Gateway:

1. Start your Web browser.
2. Download the software upgrade file from the Dialogic Support Web site.
3. In the Web browser address box, enter the IP address of the Media Gateway that you wish to upgrade.

4. When the System Login Web page appears, enter the user name and current password in the boxes provided and click on the **Log On** button.

Note: The user name and password are case sensitive.

5. Once the login has been accepted, the Media Gateway Summary Web page will appear. Select the **Upgrade** Web page from the **Configuration** menu on the left side of the page.
6. Enter the path and filename of the upgrade file or click on the **Browse...** button to select the upgrade file.
7. Click on the **Install** button to upload the file.

   The Media Gateway will respond when the upgrade has completed (approximately 10-20 seconds).
8. If there is more than one upgrade file, repeat steps 5 through 7 for each additional file.
9. Select **Restart** from the **Configuration** menu on the left side of the page.
10. On the **Restart** Web page, click on the **Restart Unit Now** or **Restart Unit When Idle** button. (See Section 2.5, “Restart Options”, on page 40).
11. The unit will now restart and re-initialize using the new software.

Parameter Reference 3

This section lists each Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) configuration parameter that may be changed using the Web browser **Configuration** menu. Also listed are non-menu (hidden) parameters which are not accessible from the Configuration menu. Changing non-menu parameters significantly modifies the operation of the Media Gateway. Included in this section is a description of each parameter, the allowed values, and, where applicable, the default value. The parameters are grouped into the following major categories:

- IP Settings 46
- Management Protocols Parameters 52
- VoIP General Parameters 59
- VoIP Network Group Parameters 80
- VoIP Media Parameters 92
- QoS Group 70
- VoIP Authentication Parameters 102
- TDM General Parameters 105
- TDM T1/E1 Parameters 109
- TDM Analog Parameters 125
- TDM Digital Parameters 135
- TDM Port Enable Parameters 136
- TDM Call Type Group 136
- TDM Cause Codes Group 141
- TDM CPID Parsing Configuration 145
- Serial Ports Parameters 146
- Serial Ports Switch Protocol Parameters 149
- Tone Detection Parameters 154
- Certificates Parameters 161
- DSP Settings Parameters 162
- Non-Menu (Hidden) Parameters 178

3.1 IP Settings

The IP Settings include the following groups:

- IP Settings, LAN1
- IP Settings, LAN2 (DMG2000)

3.1.1 IP Settings, LAN1

The IP Settings, LAN1 group includes the following parameters:

- Client IP Address
- Client Subnet Mask
- Default Network Gateway Address
- BOOTP Enabled
- SNTP Server IP Address

3.1.1.1 Client IP Address

Description: Sets the IP address of the Media Gateway.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = 10.12.13.74

INI File Parameter Name = ipClientAddr

Note: Unit requires a restart if this parameter value is changed.

3.1.1.2 Client Subnet Mask

Description: Sets the subnet mask of the Media Gateway.

Allowed Values: Any valid IP mask.

Default Value = 255.255.255.0

INI File Parameter Name = ipSubnetMask

Note: Unit requires a restart if this parameter value is changed.

3.1.1.3 Default Network Gateway Address

Description: Sets the IP address of the default network gateway router.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = blank

INI File Parameter Name = ipRouterAddr

Note: Unit requires a restart if this parameter value is changed.

3.1.1.4 BOOTP Enabled

Description: When enabled, this parameter causes the Media Gateway to issue a BOOTP request on startup. A BOOTP/TFTP server may be configured to provide IP credentials, firmware upgrades, and a configuration INI file.

Note: Either a BOOTP/TFTP server or DHCP server may be used to respond to the Media Gateway BOOTP request.

Allowed Values:

- Yes = Parameter enabled - Media Gateway issues a BOOTP request on startup.
- No = Parameter disabled - Media Gateway does not issue a BOOTP request on startup.

Default Value = Yes

INI File Parameter Name = ipBootpEnabled

Note: Unit requires a restart if this parameter value is changed.

3.1.1.5 SNTP Server IP Address

Description: Simple Network Time Protocol (SNTP) server IP address. Used for SIP Transport Layer Security (TLS) data verification

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = No default value

INI File Parameter Name = ipSntpServerAddr

Note: Unit requires a restart if this parameter value is changed.

3.1.2 IP Settings, LAN2 (DMG2000)

The IP Settings, LAN2 group includes the following parameters:

- Client IP Address
- Client Subnet Mask

Note: Currently, LAN2 is only supported in Version 5.1 SU1 Software or later.

3.1.2.1 Client IP Address

Description: Sets the IP address of the Media Gateway second Ethernet port. This Ethernet port may be used as the Maintenance port depending on the "IP Management Interface" setting. A value of 0.0.0.0 disables the LAN2 Ethernet port.

Note: The LAN2 subnet must not overlap with the LAN1 subnet other wise there will be IP routing issues.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = 0.0.0.0 (disabled)

INI File Parameter Name = ipClientAddrLan2

Note: Unit requires a restart if this parameter value is changed.

3.1.2.2 Client Subnet Mask

Description: Sets the subnet mask of the Media Gateway Maintenance Ethernet port.

Allowed Values: Any valid IP mask.

Default Value = 255.255.255.0

INI File Parameter Name = ipSubnetMaskLan2

Note: Unit requires a restart if this parameter value is changed.

3.1.3 IP Advanced Parameters (DMG2000)

The IP advanced parameters allow an advanced network topology to be configured using the LAN interfaces. The administrator can configure which LAN interfaces provide access to the IP management protocols. The administrator can also configure custom routing rules that replace or supplement the functionality of the default network gateway address.

Management Interface

The management IP protocols are accessible on all LAN interfaces by default. These management IP protocols include HTTP, HTTPS, SNMP, Syslog and Telnet. An administrator may choose to limit access to the management protocols based on the LAN interface. This means that only certain networks and subnets can access them. The management interface can be bound to LAN1 only, LAN2 only or all LANs (both LAN1 and LAN2). For example, the management protocols may be configured to be accessible only on LAN2. This means that LAN1 would not be a management interface and the management protocols would not be accessible via LAN1. The protocols would only be accessible by accessing the gateway at the LAN2 IP address.

The management interface group includes the parameter:

- IP Management Interface

3.1.3.1 IP Management Interface

Description: Sets which LAN interface is accessible for management protocols.

Allowed Values:

- LAN1 = Management functionality available on LAN1 interface only
- LAN2 = Management functionality available on LAN2 interface only
- ALL = Management functionality available on both LAN1 and LAN2 interfaces

Default Value = ALL

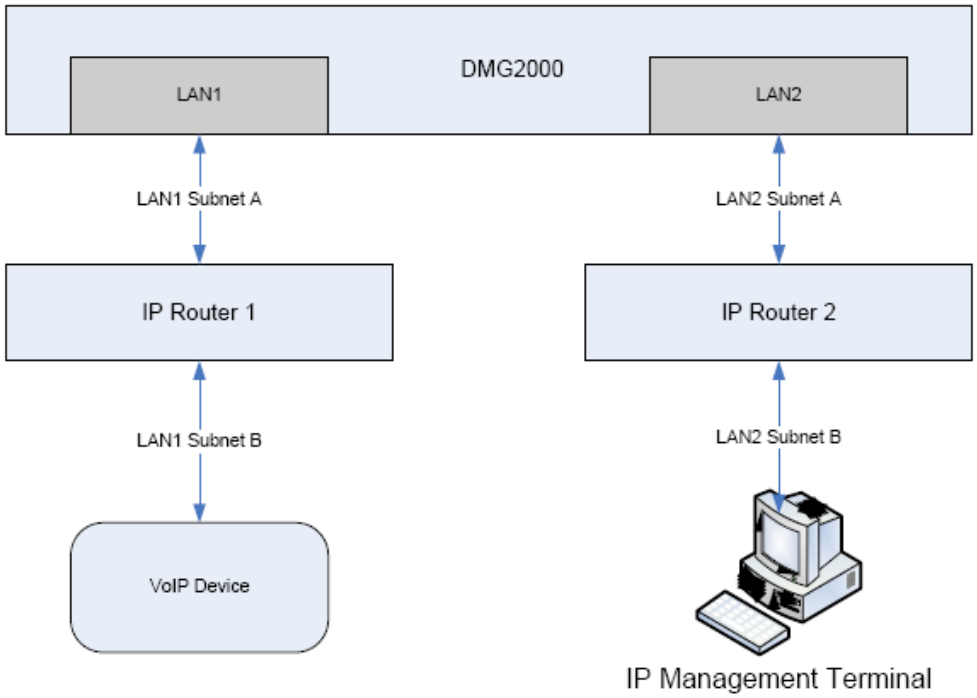
INI File Parameter Name = ipMgmt

Note: Unit requires a restart if this parameter value is changed.

3.1.3.2 Advanced IP Route Rules

A network device may only have one default network gateway address which receives transmitted IP packets that are not on the local subnet or do not match a custom routing rule. However, a single network gateway may not be sufficient or desirable in many network topologies. For example, LAN1 and LAN2 may each require their own network gateway for access. Figure 7 illustrates one such network topology.

Figure 7. Example of a Network Topology



Two routes are required in this topology so that:

1. LAN1 can access LAN1 Subnet B
2. LAN2 can access LAN2 Subnet B

The IP Router 1 may be configured as the default network gateway. However, the DMG2000 must also be configured to use IP Router 2 as a network gateway. A custom route must be created so that LAN2 can access LAN2 Subnet B. In addition, the default network gateway may be optionally replaced by another custom route. The custom route would allow LAN1 to access LAN1 Subnet B without using a default network gateway.

A default network gateway route blindly forwards IP packets whose destination is not the local subnet. A custom route is specific. It contains a single routing rule that specifically states when an IP packet is forwarded to a specific network gateway. This means that multiple custom routes may be installed on the same gateway to access multiple subnets. Each route contains a configured rule that directs the LAN interface, subnet mask and network gateway.

The advanced IP route rules include the following parameters:

- Ethernet Interface
- Destination Address
- Destination Mask
- Gateway Address

An IP route rule is a custom routing rule on a specific LAN interface. It allows access to a specific subnet that matches the destination address/mask via the specified gateway address. This rule supplements or replaces the functionality of the default network gateway address by providing access to multiple subnets across all LAN interfaces. The IP route rules are available on the DMG2000 only.

Ethernet Interface

Description: Specifies the LAN interface used for the route. The "Gateway Address" must be on the same subnet as this LAN interface.

Allowed Values:

- LAN1 = IP route will use LAN1
- LAN2 = IP route will use LAN2

Default Value = LAN1

INI File Parameter Name = ipRouteIf

Note: Unit requires a restart if this parameter value is changed.

Destination Address

Description: Destination IP address that is combined with the destination mask to define the subnet range for the route. Unused bits may be set to 0.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = blank

INI File Parameter Name = ipRouteDest

Note: Unit requires a restart if this parameter value is changed.

Destination Mask

Description: Destination IP mask that is combined with the destination address to define the subnet range for the route.

Allowed Values: Any valid IP mask

Default Value = blank

INI File Parameter Name = ipRouteMask

Note: Unit requires a restart if this parameter value is changed.

Gateway Address

Description: IP address of the gateway router that receives all IP packets from the LAN that match this route. The gateway IP address must be in the same subnet as the route's LAN interface.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = blank

INI File Parameter Name = ipRouteGw

Note: Unit requires a restart if this parameter value is changed.

3.2 Management Protocols Parameters

The Management Protocols group includes the following subgroups:

- E-Mail Group
- SysLog Group
- SNMP Group
- Web Server Group
- Telnet Server Group
- Serial Ports Group

3.2.1 E-Mail Group

The E-Mail subgroup includes the following parameters:

- E-Mail Alarms Enabled
- E-Mail Minimum Alarm Severity
- Destination E-Mail List
- E-Mail Server IP Address
- Source E-Mail Address

E-Mail Alarms Enabled

Description: Defines whether an e-mail is generated to signal an alarm.

Allowed Values:

- Yes = E-mail is generated.
- No = E-mail is not generated.

Default Value = No

INI File Parameter Name = emailEnabled

E-Mail Minimum Alarm Severity

Description: Defines the minimum alarm severity level that generates an e-mail notification.

Allowed Values:

- Error
- Warning
- Info

Default Value = Info

INI File Parameter Name = emailAlarmSeverity

Destination E-Mail List

Description: Semi-colon delimited list of e-mail addresses that will receive e-mail alarms.

Allowed Values: Semi-colon delimited list of e-mail addresses.

Default Value = (no default value)

INI File Parameter Name = emailList

E-Mail Server IP Address

Description: IP addresses of the SMTP e-mail server that receive the e-mail generated by the IP gateway.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = emailServerAddr

Source E-Mail Address

Description: E-mail address used as source of e-mail alarm messages.

Allowed Values: Any valid e-mail address

Default Value = alarm@pbxgw.com

INI File Parameter Name = emailSource

3.2.2 SysLog Group

The SysLog subgroup includes the following parameters:

- SysLog Server IP Address
- Alarms to SysLog Enabled
- SysLog Minimum Alarm Severity
- Diagnostics Trace to SysLog Enabled

SysLog Server IP Address

Description: IP address of the SysLog (RFC3164) server (notifications sent to server's UDP port 514).

Allowed Values: Any valid IP address in dotted decimal format

Default Value = (no default value)

INI File Parameter Name = syslogServerAddr

Note: Unit requires a restart if this parameter value is changed.

Alarms to SysLog Enabled

Description: Defines whether alarms are sent to the SysLog server.

Allowed Values:

- Yes = Alarms are sent as user-level facility messages.
- No = Alarms are sent to SysLog server.

Default Value = No

INI File Parameter Name = syslogAlarmEnabled

SysLog Minimum Alarm Severity

Description: Defines the minimum alarm severity level that generates notifications to SysLog server.

Allowed Values:

- Error = Error alarms are sent to the SysLog server.
- Warn = Error and Warning alarms are sent to the SysLog server.
- Info = Error, Warning, and Informational alarms are sent to the SysLog server.

Default Value = Info

INI File Parameter Name = syslogAlarmSeverity

Diagnostics Trace to SysLog Enabled

Description: Defines whether diagnostic trace messages are sent to SysLog server.

Allowed Values:

- Yes = Diagnostics trace messages are sent as user-level facility/debug messages.
- No = Diagnostics trace messages are not sent as user-level facility/debug messages.

Default Value = No

INI File Parameter Name = syslogTraceEnabled

3.2.3 SNMP Group

The SNMP subgroup includes the following parameters:

- SNMP Traps Enabled?
- SNMP Minimum Alarm Severity
- SNMP Trap IP List
- SNMP Community Name
- SNMP System Name
- SNMP System Contact
- SNMP System Location

SNMP Traps Enabled?

Description: Defines whether an SNMP trap is generated to signal an alarm.

Allowed Values:

- Yes = SNMP trap is generated.
- No = SNMP trap is not generated.

Default Value = No

INI File Parameter Name = snmpTrapEnabled

SNMP Minimum Alarm Severity

Description: Defines the minimum alarm severity level to generate an SNMP trap.

Allowed Values:

- Error
- Warning
- Info

Default Value = Info

INI File Parameter Name = snmpAlarmSeverity

SNMP Trap IP List

Description: Semi-colon delimited list of the IP addresses of SNMP Managers who are to receive SNMP traps generated by the IP gateway.

Allowed Values: Semi-colon delimited list of IP addresses.

Default Value = 255.255.255.255

INI File Parameter Name = snmpTrapAddressList

SNMP Community Name

Description: Specifies the SNMP Community name. The specified community has read-only capabilities.

Allowed Values: Any string with length between 5 - 14 characters

Default Value = public

INI File Parameter Name = snmpCommunity

Note: Unit requires a restart if this parameter value is changed.

SNMP System Name

Description: Specifies the SNMP System name.

Allowed Values: Any string with length between 0 - 63 characters

Default Value = (no default value)

INI File Parameter Name = snmpSysName

Note: Unit requires a restart if this parameter value is changed.

SNMP System Contact

Description: Specifies the SNMP System administration contact name.

Allowed Values: Any string with length between 0 - 63 characters

Default Value = (no default value)

INI File Parameter Name = snmpSysContact

Note: Unit requires a restart if this parameter value is changed.

SNMP System Location

Description: Specifies the SNMP System location.

Allowed Values: Any string with length between 0 - 63 characters

Default Value = (no default value)

INI File Parameter Name = snmpSysLocation

Note: Unit requires a restart if this parameter value is changed.

3.2.4 Web Server Group

The Web Server subgroup includes the following parameters:

- HTTP Server Enabled
- HTTPs Server Enabled

HTTP Server Enabled

Description: Defines whether the HTTP Server is enabled or disabled. Disable the HTTP Server to disable non-secure Web access.

Allowed Values:

- Yes = HTTP Server is enabled.
- No = HTTP Server is disabled.

Default Value = Yes

INI File Parameter Name = webHttpEnabled

Note: Unit requires a restart if this parameter value is changed.

HTTPs Server Enabled

Description: Defines whether the HTTPs Server is enabled or disabled. Disable the HTTP Server to disable non-secure Web access.

Allowed Values:

- Yes = HTTPs Server is enabled.
- No = HTTPs Server is disabled.

Default Value = No

INI File Parameter Name = webHttpsEnabled

Note: Unit requires a restart if this parameter value is changed.

3.2.5 Telnet Server Group

The Telnet Server subgroup includes the following parameter:

- Telnet Server Enabled

Telnet Server Enabled

Description: Defines whether the Telnet Server is enabled. Disable the Telnet Server to prevent any Telnet connections to the Media Gateway.

Allowed Values:

- Yes
- No

Default Value = Yes

INI File Parameter Name = telnetEnabled

Note: Unit requires a restart if this parameter value is changed.

3.2.6 Serial Ports Group

The Serial ports subgroup includes the following parameter:

- Maintenance Port Enabled

Maintenance Port Enabled

Description: Defines whether the Maintenance port is enabled. Disable to prevent any Maintenance messages from being transmitted by the port. If disabled, the Maintenance port will show only boot messages before the port is disabled.

Note: If the Web interface and Maintenance port are both turned off, the only way to re-enable the Maintenance port is to re-enable the port through booter as follows:

1. Cycle power on unit.
2. Break into the booter by typing "pimg" immediately when prompted with the following: "Type pimg to stop auto-boot..." A direct Serial connect between the Gateway and PC should be used.

   Note: There is 2 seconds to start typing and 5 seconds to finish.
3. At the BOOT> prompt, type "pwd" command to login and then enter password.
4. At the BOOT-admin> prompt, type "reset" command.
5. The Gateway now resets to factory defaults; one of which is Serial port enabled. The Gateway will then continue to start the Gateway application.
   Caution: IP address is also reset so the user must restore it before the Web interface will work.

Allowed Values:

- Yes
- No

Default Value = Yes

INI File Parameter Name = maintPortEnabled

Note: Unit requires a restart if this parameter value is changed.

3.3 VoIP General Parameters

The VoIP General parameters include the following groups:

- User-Agent Group
- Server Group
- TCP/UDP Group
- TLS Group
- Timing Group
- Monitoring Group
- QoS Group
- Privacy Headers Group

3.3.1 User-Agent Group

The User-Agent subgroup includes the following parameters:

- Host and Domain Name
- User-Agent Header Value
- Call as Domain Name?
- Invite Expiration
- Reliable Provisional Responses

3.3.1.1 Host and Domain Name

Description: The host and domain name of the Media Gateway. This name is used when the unit registers with the SIP Registration Server.

Allowed Values: Domain Name string

Default Value = pbxgw.default.com

INI File Parameter Name = sipServerDomain

Note: Unit requires a restart if this parameter value is changed.

3.3.1.2 User-Agent Header Value

Description: Defines User-Agent header value that will be used in SIP requests.

Allowed Values: Any string with length between 1 - 29 characters

Default Value = (no default value)

INI File Parameter Name = sipUserAgentHdr

3.3.1.3 Call as Domain Name?

Description: Defines the host name used in the From header of generated INVITE requests and the Contact field.

Allowed Values:

- Yes = The Media Gateway's domain name is used as the Host Name.
- No = The Media Gateway's IP address is used as the Host Name.

Default Value = No

INI File Parameter Name = sipCallAsDomainName

3.3.1.4 Invite Expiration

Description: Specifies the amount of time in seconds that an INVITE request sent by the Media Gateway is valid and can be accepted by the SIP endpoint. After the defined time in seconds, the INVITE request expires and is no longer valid.

Allowed Values: 1 - 60000 seconds

Default Value = 60000 seconds

INI File Parameter Name = sipExpInvSec

3.3.1.5 Reliable Provisional Responses

Description: Specifies whether the reliability of provisional responses (PRACK) is disabled, supported, or required by the UAC. This is signaled with 100rel in the "Supported" and "Required" SIP headers.

Allowed Values:

- None = Reliable Provisional Response support is disabled.
- Supported = Reliable Provisional Response is supported. The INVITE messages generated by the gateway will contain 100rel in the Supported header sent to the VoIP endpoint.
- Required = Reliable Provisional Response is required. The INVITE messages generated by the gateway will contain 100rel in the Required header sent to the VoIP endpoint.

Default Value = Supported

INI File Parameter Name = sipRelProvRsp

3.3.2 Server Group

The Server subgroup includes the following parameters:

- DNS Server Address
- DNS Server Address 2
- DNS Translation of Phone Numbers

3.3.2.1 DNS Server Address

Description: Specifies the IP address of the Domain Name Server (DNS) that the Media Gateway will use to resolve IP address information.

Notes: *1.* If the DNS Server IP Address parameter is configured but the Primary Proxy Server Address parameter is not, the Media Gateway will use the DNS Server to resolve IP address information.

2. If both the DNS Server IP Address parameter and the Primary Proxy Server Address parameter are configured, the Media Gateway will use the Proxy Server for all requests.

3. If the DNS Server IP Address parameter is configured and the Primary Proxy Server Address parameter is configured with an alias, then the Media Gateway will first use the DNS Server to resolve the alias of the Proxy Server to an IP address and then use the Proxy Server for all subsequent requests.

Allowed Values: Any valid IP Address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = sipDnsServerAddr

Note: Unit requires a restart if this parameter value is changed.

3.3.2.2 DNS Server Address 2

Description: Specifies the IP address of the Domain Name Server (DNS) that the Media Gateway will use to resolve IP address information.

Allowed Values: Any valid IP Address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = sipDnsServerAddr2

Note: Unit requires a restart if this parameter value is changed.

3.3.2.3 DNS Translation of Phone Numbers

Description: If ‘Yes’, the Media Gateway will use DNS to translate tel URIs and URIs with the 'user=phone' parameter. If ‘No’, the Media Gateway will not use DNS to translate these types of URIs.

Allowed Values:

- Yes
- No

Default Value = No

INI File Parameter Name = sipEnumDnsEnabled

3.3.3 TCP/UDP Group

The TCP/UDP subgroup includes the following parameters:

- UDP/TCP Transport Enabled
- TCP/UDP Server Port
- TCP Inactivity Timer

3.3.3.1 UDP/TCP Transport Enabled

Description: Enables or disables the UDP/TCP transports.

Allowed Values:

- Yes = UDP and TCP transports are enabled.
- No = UDP and TCP transports are disabled.

Default Value = Yes

INI File Parameter Name = sipUdpTcpEnabled

Note: Unit requires a restart if this parameter value is changed.

3.3.3.2 TCP/UDP Server Port

Description: The TCP/UDP Port of the Media Gateway on which SIP messages are sent/received.

Allowed Values: 1024 - 65000

Default Value = 5060

INI File Parameter Name = sipServerPort

Note: Unit requires a restart if this parameter value is changed.

3.3.3.3 TCP Inactivity Timer

Description: Number of seconds after which an idle Transmission Control Protocol (TCP) connection will be closed.

Allowed Values: 10 - 60000 seconds

Default Value = 90 seconds

INI File Parameter Name = sipTcpInactivitySec

3.3.4 TLS Group

The TLS subgroup includes the following parameters:

- TLS Transport Enabled
- TLS Server Port
- SSL TLS Protocol
- Mutual TLS Authentication Required
- TLS Inactivity Timer
- Verify TLS Peer Certificate Date
- Verify TLS Peer Certificate Trust
- Verify TLS Peer Certificate Purpose

3.3.4.1 TLS Transport Enabled

Description: Determines whether the TLS transport is enabled or disabled.

Allowed Values:

- Yes = TLS transport is enabled.
- No = TLS transport is disabled.

Default Value = No

INI File Parameter Name = sipTlsEnabled

Note: Unit requires a restart if this parameter value is changed.

3.3.4.2 TLS Server Port

Description: The Transport Layer Security (TLS) Port of the Media Gateway on which SIP TLS messages are sent/received.

Allowed Values: 1024 - 65000

Default Value = 5061

INI File Parameter Name = sipTlsServerPort

Note: Unit requires a restart if this parameter value is changed.

3.3.4.3 SSL TLS Protocol

Description: Specifies the default SSL record type to be used on TLS connections.

Allowed Values:

- SSLv3_TLSv1 = Connection will understand both the SSLv3 and TLSv1 protocols.
- SSLv3_Only = Connection will only understand the SSLv3 protocol.
- TLSv1_Only = Connection will only understand the TLSv1 protocol.

Default Value = SSLv3_TLSv1

INI File Parameter Name = secSipTlsProtocol

Note: Unit requires a restart if this parameter value is changed.

3.3.4.4 Mutual TLS Authentication Required

Description: If 'Yes' is selected, the SIP client sends a certificate to the peer server and the SIP server will send a certificate request to the client. The certificate received from the client is validated. If the client does not provide a valid certificate, the server will close the session.

If 'No' is selected, the SIP server does not send a client certificate request to the client and the client does not send a certificate to the server. A certificate received from the client is ignored by the server.

Allowed Values:

- Yes = SIP client sends certificate to peer server and SIP server sends certificate request to client.
- No = SIP server does not send client a certificate request and client does not send a certificate to the server.

Default Value = Yes

INI File Parameter Name = sipTlsMutualAuthentication

Note: Unit requires a restart if this parameter value is changed.

3.3.4.5 TLS Inactivity Timer

Description: Number of seconds after which an idle Transport Layer Security (TLS) connection will be closed.

Allowed Values: 10 - 60000 seconds

Default Value = 30 seconds

INI File Parameter Name = sipTlsInactivitySec

3.3.4.6 Verify TLS Peer Certificate Date

Description: Determines whether the TLS peer's certificate date is validated. This requires that the Media Gateway has already received the network time via SNTP.

Allowed Values:

- Yes = Validate certificate date
- No = Do not validate certificate date

Default Value = Yes

INI File Parameter Name = sipTlsCertVerifyDate

3.3.4.7 Verify TLS Peer Certificate Trust

Description: Determines whether the TLS peer's certificate trust relationship is validated against the list of trusted CA certificates.

Allowed Values:

- Yes = Validate
- No = Do not validate

Default Value = Yes

INI File Parameter Name = sipTlsCertVerifyTrust

3.3.4.8 Verify TLS Peer Certificate Purpose

Description: Determines if the purpose of the TLS Peer Certificate should be verified.

Allowed Values:

- Yes = Validate
- No = Do not validate

Default Value = Yes

INI File Parameter Name = sipTlsCertVerifyPurpose

3.3.5 Timing Group

The Timing subgroup includes the following parameters:

- T1 Time
- T2 Time
- T4 Time
- T1 Multiplier

3.3.5.1 T1 Time

Description: The T1 Time specifies the SIP request retransmit timeout in milliseconds. This timer is started when a SIP request is generated. If no response to the request is received in T1 Time milliseconds, the request is retransmitted and the timeout is doubled to 2*T1 Time milliseconds. If again no response is received before the new timeout, the message is again retransmitted and the timeout is again doubled - this time to 4*T1 Time milliseconds.

Allowed Values: 200 - 60000 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = sipT1TimeMs

3.3.5.2 T2 Time

Description: The T2 Time specifies the maximum retransmit time of SIP request messages (except for INVITE) in milliseconds. The retransmitting of the request and the doubling of the timeout continues until the timeout reaches T2 Time milliseconds. Once the timeout reaches T2 Time, the request is retransmitted at T2 Time intervals until a response is received or until the message has been retransmitted "NumRetriesRequest" times - at which point the request expires.

Allowed Values: 200 - 60000 milliseconds

Default Value = 4000 milliseconds

INI File Parameter Name = sipT2TimeMs

3.3.5.3 T4 Time

Description: The T4 Time specifies the time in milliseconds that the network will take to clear messages between client and server transactions.

Allowed Values: 1000 - 60000 milliseconds

Default Value = 5000 milliseconds

INI File Parameter Name = sipT4TimeMs

3.3.5.4 T1 Multiplier

Description: Specifies the value to be multiplied by the "T1 Timer" parameter to determine the timeout for a SIP request. For example, if the T1 Timer is set to 500 milliseconds and the T1 Multiplier is set to 64, then the SIP request will fail in 32 seconds (64 * 0.5 seconds).

Allowed Values: 1 - 255

Default Value = 64

INI File Parameter Name = sipT1Multiplier

Note: Unit requires a restart if this parameter value is changed.

3.3.6 Monitoring Group

Note: The Monitoring subgroup parameters only apply to a Media Gateway operating in the Phone Emulating Mode.

The Monitoring subgroup includes the following parameters:

- Monitor Call Connections
- Call Monitor Interval
- VoIP Host Monitor Interval
- Proactive DNS Monitoring

Monitor Call Connections

Description: When the Monitor Call Connections parameter is enabled, the Media Gateway will monitor the connection state of active IP calls. If the active IP call has lost connection, the Media Gateway will tear down the call.

Note: If enabled in SIP, the IP endpoints must support session timers.

Allowed Values:

- Yes = Media Gateway monitors the connection state of active IP calls.
- No = Media Gateway does not monitor the connection state of active IP calls.

Default Value = No

INI File Parameter Name = gwMonitorCallConns

Call Monitor Interval

Description: Specifies the call monitor interval in seconds at which active IP calls are monitored in order to determine the connections status. This parameter is only valid if the Monitor Call Connections parameter is enabled.

Allowed Values: 30 through 3600 seconds

Default Value = 60

INI File Parameter Name = gwMonitorCallIntSec

VoIP Host Monitor Interval

Description: Interval in which VoIP hosts are monitored to determine on-line status. Valid only if there are VoIP Host Groups in the Routing Table that have Fault Tolerance enabled.

Allowed Values: 10 - 3600

Default Value = 30

INI File Parameter Name = gwMonitorVoipHostsIntSec

Note: Unit requires a restart if this parameter value is changed.

Proactive DNS Monitoring

Description: When Proactive DNS Monitoring is enabled, the gateway will monitor the IP addresses obtained from DNS resolution. If an IP address does not respond to SIP OPTIONS pings, the gateway does not attempt to send it SIP INVITES. When the IP address resumes responding to SIP OPTIONS pings, the gateway resumes sending it SIP INVITES.

Allowed Values:

- Yes = the gateway monitors the state of DNS resolved IP addresses.
- No = the gateway does not monitor the state of DNS resolved IP addresses.

Default Value = No

INI File Parameter Name = gwProactiveMonitorDnsARecordsEnabled

Note: Unit requires a restart if this parameter value is changed.

3.3.7 QoS Group

The Quality of Service (QoS) subgroup includes the following parameters:

- Call Control QoS Byte

Call Control QoS Byte

Description: The Call Control QoS Byte parameter defines a decimal byte value that represents QoS bit flags. This parameter is used in each call control (SIP) data packet transmitted from the Media Gateway with a QoS byte code. Routers use this byte code to assign priority levels to packets. The QoS byte may be interpreted as either IPv4 TOS or DiffServ.

IPv4 TOS Byte

Bits:
0-2 = Precedence (RFC 1122)
3-6 = Type of Service (RFC 1349/1455)
7 = Must be zero

PrecedenceBits (0-2):
111xxxxx (224) = Network Control
110xxxxx (192) = Internetwork Control
101xxxxx (160) = CRITIC/ECP
100xxxxx (128) = Flash Override
011xxxxx (96) = Flash
010xxxxx (64) = Immediate
001xxxxx (32 = Priority
000xxxxx (0) = Routine

Type of Service Bits (3-6):
xxx0000x (0) = Normal
xxx1000x (16) = Minimize delay
xxx0100x (8) = Maximize throughput
xxx0010x (4) = Maximize reliability
xxx0001x (2) = Minimize monetary cost
xxx1111x (30) = Maximize physical link security
xxx (32 = Priority
xxx (0) = Routine

Default Value = 0 (Routine/Normal)

DiffServ Codepoint Byte

Bits:
0-5 = Differentiated Services Codepoint (RFC 2474)
6-7 = Explicit Congestion Notification (RFC 2481)

Allowed Values: 0-255

Default Value = 0

INI File Parameter Name = gwQosCallControl

3.3.8 Privacy Headers Group

Note: The Privacy Headers parameters only apply to a DMG2000 that is configured for SIP (on the VoIP side) and ISDN (on the TDM side).

Privacy Headers are used in SIP messages to specify to the receiving side how to present 'Calling Party ID' data to a user. When privacy is enabled, the normal 'From' fields in a SIP message are masked and the actual 'Calling Party ID' is placed in 'privacy headers'. An example is shown following:

From: "Anonymous" sip:anonymous@anonymous.invalid

P-Asserted-Identity: "Mary Doe" sip:1001@192.168.1.30

In the example, the normal data in the 'From' fields are masked (with anonymous) and the actual data are placed in the 'P-Asserted-Identity' fields. The 'From' fields are presented to the user, while the 'P-Asserted-Identity' fields are used only for routing.

The ISDN protocol contains fields analogous to the SIP privacy headers.

The Privacy Headers subgroup includes the following parameters:

- Enable Privacy Headers
- Privacy Header Method

Enable Privacy Headers

Description: When Privacy Headers are enabled, the gateway translates received SIP privacy headers to transmitted ISDN privacy fields and vice versa. When disabled, the gateway ignores any received SIP privacy headers or any received ISDN privacy fields.

Allowed Values:

- Yes = Enables translation between SIP Privacy Headers and ISDN privacy fields.
- No = Disables translation between SIP Privacy Headers and ISDN privacy fields.

Default Value = No

INI File Parameter Name = sipPrivacyHdrEnabled

Privacy Header Method

Description: The SIP protocol provides for two methods to keep the 'Calling Party ID' data private. Both methods modify the 'From' header by changing the fields to 'anonymous'. They differ on how the actual 'Calling Party ID' data is presented. The first method is shown in the preceding example, that is, P-Asserted-Identity' is used. The second method uses 'Remote-Party-ID'. Both methods supply the same data, just in somewhat different formats.

Note: Privacy Header Method is relevant only when Enable Privacy Headers is set to 'yes'.

Allowed Values:

- P-Asserted-Identification = When a SIP 'P-Asserted-Identity' header is received its data is translated into ISDN privacy fields. When ISDN privacy fields are received they are converted into a SIP 'P-Asserted-Identity' header. If a SIP 'Remote-Party-ID' header is received it is ignored.
- Remote-Party-ID = When a SIP 'Remote-Party-ID' header is received its data is translated into ISDN privacy fields. When ISDN privacy fields are received they are converted into a SIP 'Remote-Party-ID' header. If a 'P-Asserted-Identity' header is received it is ignored.
- Both = This setting is used when some SIP devices communicating with the gateway use 'P-Asserted-Identity' for privacy and others use 'Remote-Party-ID' for privacy. So when either SIP method is received the gateway converts into ISDN privacy fields. When ISDN privacy fields are received the gateway creates a SIP message that contains both a 'P-Asserted-Identity' header and a 'Remote-Party-ID' header. The receiving party will use the header it supports.

Default Value = P-Asserted-Identification

INI File Parameter Name = sipPrivacyHdrMethod

3.3.9 User-User Information Group

The User-User Information (UUI) parameters control the translation of information between ISDN user-user IEs and SIP user-to-user headers. The ISDN user-user IE is specified in Q.931.

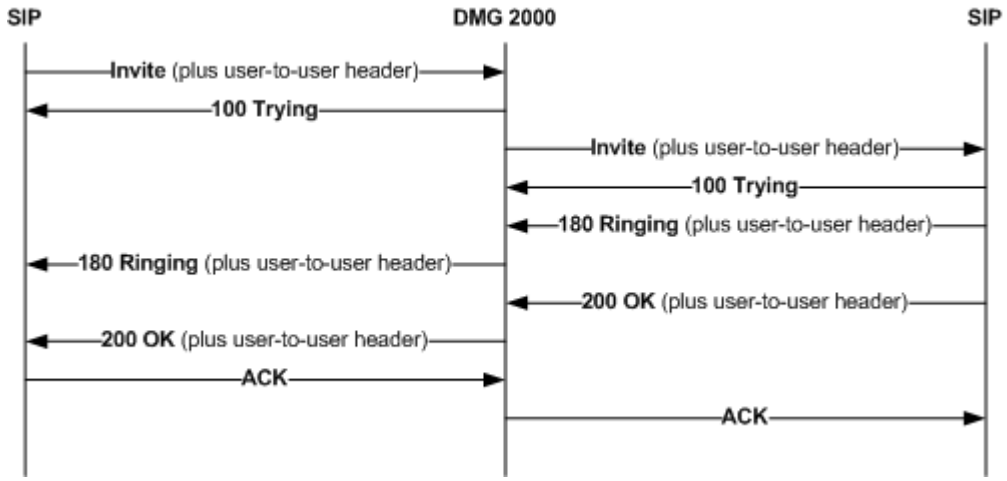
The DMG2000 supports sending UUI between called and calling parties. The purpose of UUI is to pass specific information not pertaining to the call from one user to another. The information is transferred through the gateway and does not have any effect on the call. The gateway supports UUI for SIP-SIP calls, SIP-ISDN calls, ISDN-SIP calls, and transfers that use SIP REFER.

SIP-SIP Calls

The UUI header is passed unchanged between calling and called parties. The gateway supports a single UUI header in the 'INVITE', '180 Ringing' and '200 OK' messages.

Figure 8 is an example call flow for SIP-SIP calls:

Figure 8. Example Call Flow for SIP-SIP Calls

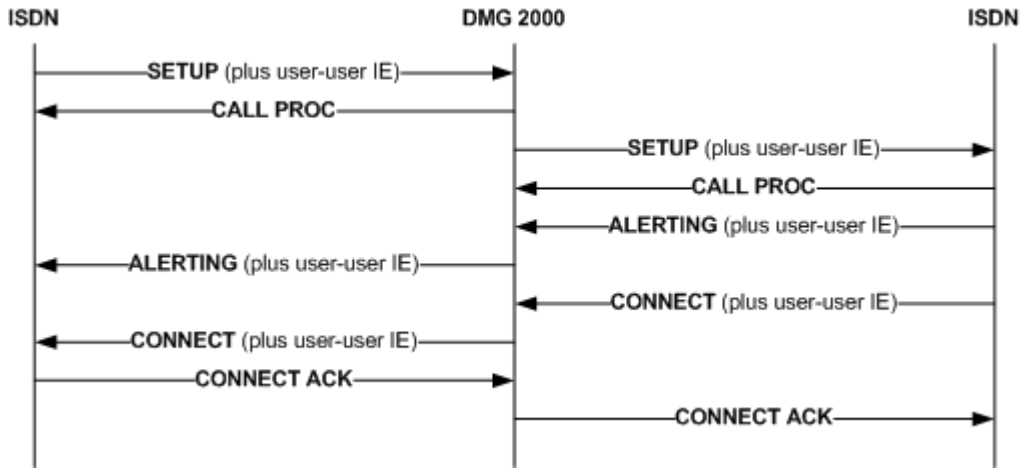


ISDN-ISDN Calls

The UUI information element is passed unchanged between calling and called parties. The gateway support a single UUI information element in the 'SETUP', 'ALERTING' and 'CONNECT' messages.

Figure 9 is an example call flow for ISDN-ISDN calls:

Figure 9. Example Call Flow for ISDN-ISDN Calls



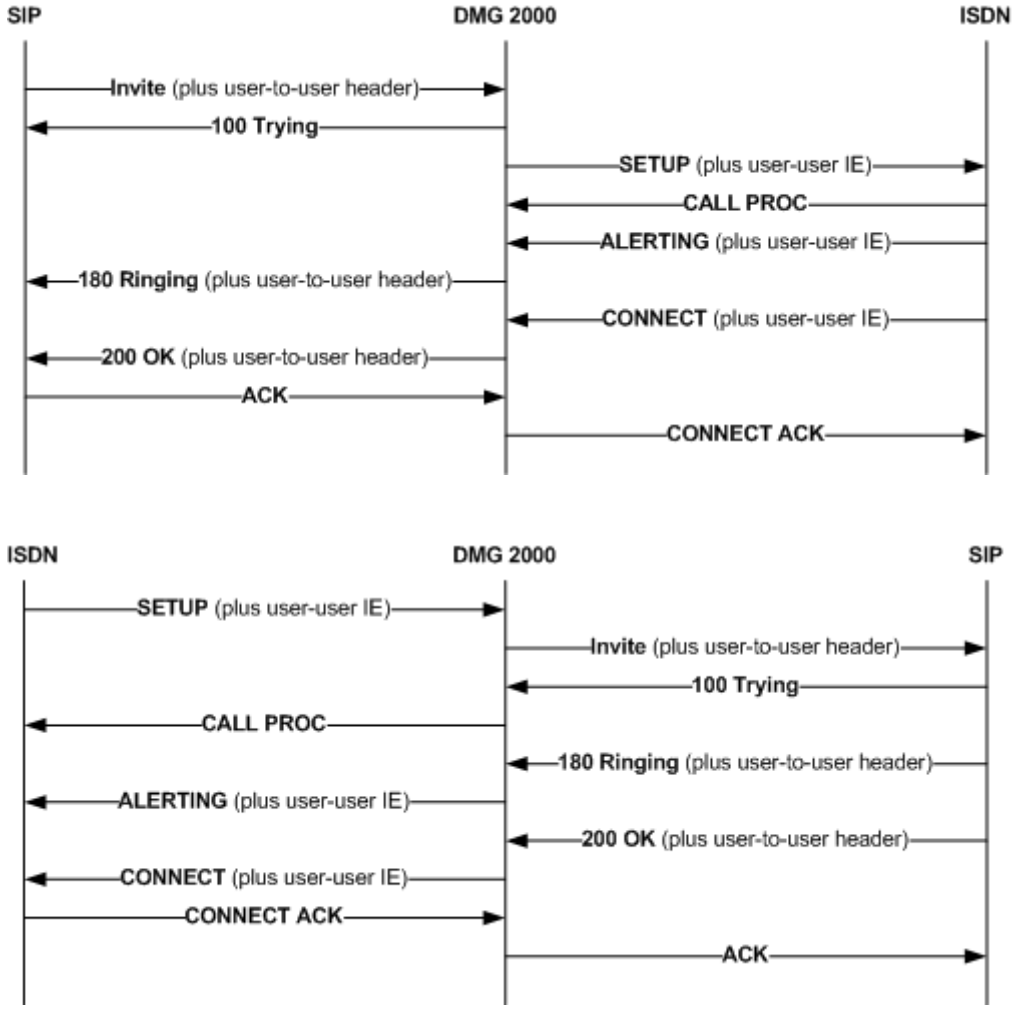
SIP-ISDN and ISDN-SIP Calls

For calls between SIP and ISDN, the UUI is translated between formats. SIP UUI headers are translated to ISDN UUI information elements and ISDN UUI information elements are translated to SIP UUI headers.

Translation occurs only for specific message pairs. Translation occurs between SIP 'INVITE' and ISDN 'SETUP', between SIP '180 Ringing' and ISDN 'ALERTING', and between SIP '200 OK' and ISDN 'CONNECT'.

Figure 10 is an example call flow for SIP-ISDN and ISDN-SIP calls:

Figure 10. Example Call Flow for SIP-ISDN and ISDN-SIP Calls

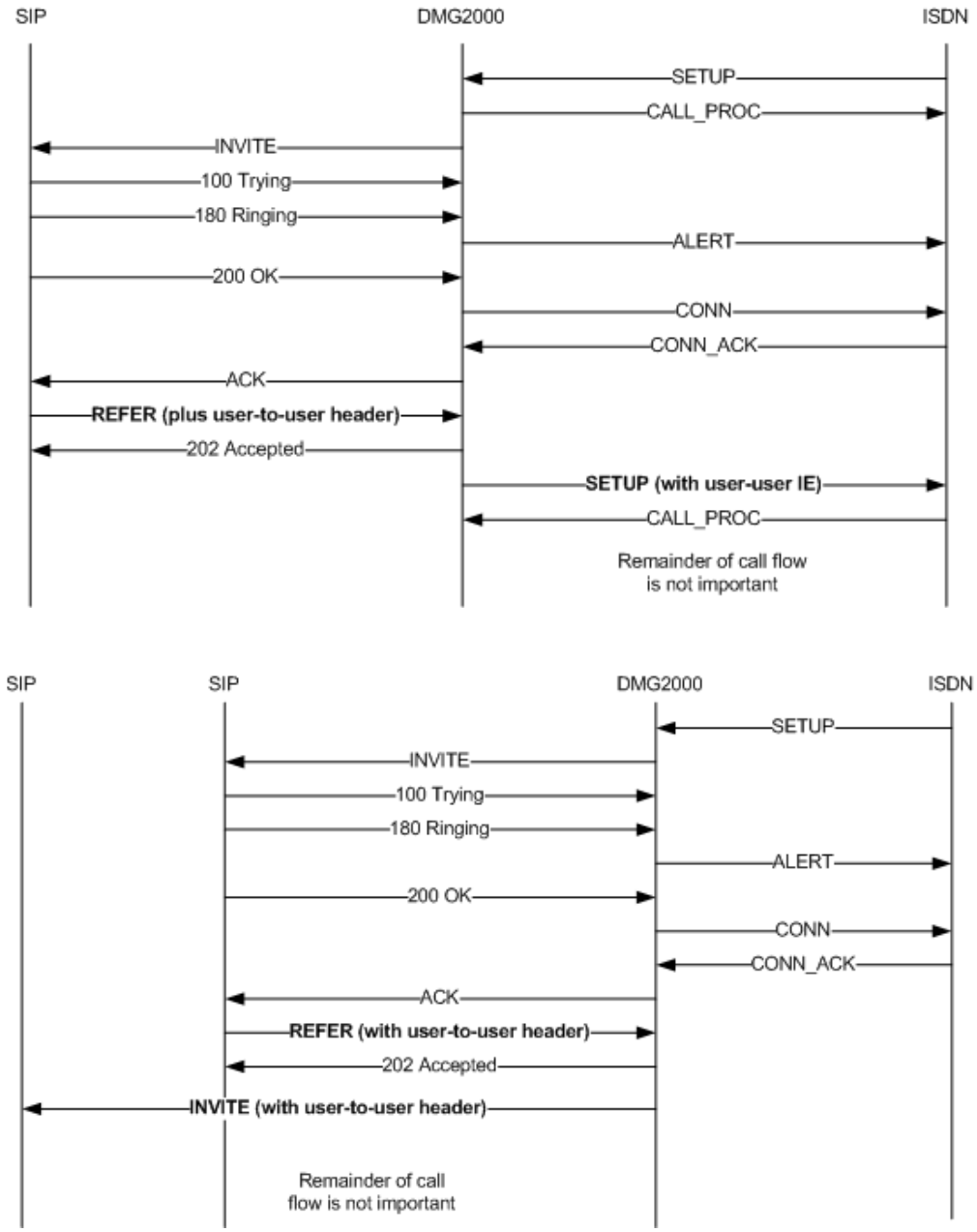


Transfers using SIP REFER

When the DMG2000 receives a SIP REFER that contains UUI, the UUI is passed to the resulting SIP INVITE or ISDN SETUP.

Figure 11 is an example call flow for transfers using SIP REFER:

Figure 11. Example Call Flow for Transfers using SIP REFER



The User-User Information subgroup includes the following parameters:

- User-User Enabled
- Translation Method

User-User Enabled

Description: Specifies how the DMG2000 processes SIP UUI headers and ISDN UUI information elements are processed by the DMG 2000.

Allowed Values:

- Yes = The SIP UUI headers and ISDN UUI information elements are processed.
- No = The SIP UUI headers and ISDN UUI information elements are ignored.

Default Value = No

INI File Parameter Name = gwU2UEnable

Translation Method

Description: Specifies how to translate between ISDN user-user information and SIP user-user headers. For inbound SIP messages, this parameter is used only with the user-user header that does not explicitly specify the method.

Allowed Values:

- Hex
  - ISDN to SIP - Each byte is translated into two ASCII hex digits.
  - SIP to ISDN - Two ASCII hex digits are translated into one byte.
- String
  - ISDN to SIP - Each byte is translated into a single ASCII character. Any bytes that don't translate into displayable characters are ignored.
  - SIP to ISDN - Each ASCII character is translated into a single byte.

Default Value = String

INI File Parameter Name = gwU2UTranslateMethod

3.3.10 ISUP Using Multipart MIME Group

The ISUP Using Multipart MIME parameters control the propagation of ISDN information elements to SIP messages. The information elements are, typically, proprietary to a telephony vendor and do not map to any defined SIP header. The DMG2000 uses multipart MIME bodies to encapsulate the information element's data. Various header fields allow the SIP endpoint to decode the MIME body.

The following is an example of using multipart MIME to send an ISDN information element. The line numbers are not part of the message; they are used in the discussion following the example.

```
1   INVITE sip:4700@10.10.11.47 SIP/2.0
2   From:<sip:2042@10.10.11.114:5060;user=phone>;vnd.pimg.port=23;tag=5A533246
3   To:<sip:4700@10.10.11.47>
4   Contact:<sip:2042@10.10.11.114:5060>
5   Mime-Version: 1.0
6   Content-Type:multipart/mixed ;boundary=”?(boundary)*”
7   Supported:replaces,early-session,100rel
8   Allow:INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,PRACK,REFER,NOTIFY
9   Expires:120
10  Call-ID:01B246A660AA00460000000E@pbxgw.default.com
11  CSeq:1 INVITE
12  Max-Forwards:70
13  User-Agent:PBX-IP Media Gateway
14  Via:SIP/2.0/UDP 10.10.11.114:5060;branch=z9hG4bKE194EDF
15  Content-Length:291
16
17  --?(boundary)*
18  Content-Type: application/SDP
19  v=0
20  o=phone 15122 16488 IN IP4 10.10.11.114
21  s=-
22  c=IN IP4 10.10.11.114
23  t=0 0
24  m=audio 49030 RTP/AVP 0 13
25  a=rtpmap:0 PCMU/8000/1
26  a=ptime:30
27  a=rtpmap:13 CN/8000
28
29  --?(boundary)*
30  Content-Type: application/ISUP; version=Dialogic-DMG2000-v1
31  Content-Disposition: signal ; handling=optional
32  Content-Description: ISDN IE 77h
33
34  37 31 36 35 35 35 36 35 35 32
35
36  --?(boundary)*--
```

Lines 5 and 6 specify that multipart MIME is appended to the INVITE message and that it uses MIME version 1.0. Line 6 defines the boundary between MIME bodies; in our example the boundary is the string ?(boundary)*.

Line 17 starts the first MIME body. This body sends SDP (session description protocol) to the SIP endpoint. We are not concerned with this.

Line 29 starts the second MIME body. This body sends the ISDN information element's data. Lines 30-32 describe the body to allow the SIP endpoint to decode it. Line 30 states the MIME body version. Line 31 says that processing the MIME body is optional. This allows for the case when some SIP endpoints need the ISDN information element but others don't. Line 32 specifies the content of body as an ISDN information element number 77 (hex). Each data byte of the information element is translated into two hex digits, line 34. Line 36 simultaneously ends the second body and the multipart MIME.

France Telecom's IE77

Description: Allows propagating an ISDN Information 77 (hex) from an ISDN SETUP message to a SIP INVITE.

Allowed Values:

- Yes = Information Element 77 (hex) is included in a SIP INVITE message.
- No = Information Element is ignored.

Default Value = No (preserves pre Version 6.0 SU8 Software behavior)

INI File Parameter Name = gwIsdnIe77Enable

Format of MIME body that encapsulates Information Element 77 (hex)

```
1  Content-Type: application/ISUP; version=Dialogic-DMG2000-v1
2  Content-Disposition: signal ; handling=optional
3  Content-Description: ISDN IE 77h
4
5  37 31 36 35 35 35 36 35 35 32
6
```

Line 5 is the IE's data. Each byte within the IE is translated into two ascii hex digits.

3.4 VoIP Network Group Parameters

The VoIP Network Group feature on the gateway provides a means for the gateway to communicate to more than one type of VoIP Network. Specifically, the user can configure up to four unique VoIP Networks - each having their own specific VoIP attributes. This allows the gateway to not only communicate with more than one type of VoIP Network on TDM-to-IP or IP-to-TDM calls, but also allows the gateway to transcode calls between two disparate VoIP networks on VoIP-to-VoIP Calls.

The VoIP Network Group parameters include the following subgroups:

- Network Group
- Transport Group
- URI Parameters Group
- Proxy Group
- Registration Group
- Audio Group
- SRTP Group

Figure 12 is a screen shot of the VoIP Network Group Web page:

Figure 12. VoIP Network Group Web Page

VoIP Network Group Configuration	
Network Group	
Network Group Label	Network Group #1
Transport	
Transport Protocol	UDP
SIPS URI Scheme	No
URI Parameters	
User Phone Parameter	Yes
Local Phone Context	
Remote Phone Context	
Proxy	
Primary Proxy Server Address	
Primary Proxy Server Port	5060
Backup Proxy Server Address	
Backup Proxy Server Port	5060
Proxy Query Interval (sec)	2147483647
Registration	
Registration Server Address	
Registration Server Port	5060
Registered User	bill
Gateway Name	
Registration Expiration (sec)	3600
Audio	
Codec #1	G.711u
Codec #2	G.711a
Codec #3	G.722
Codec #4	None
G.711 Packet Size (ms)	20
G.723.1 Packet Size (ms)	30
G.729AB Packet Size (ms)	10
SRTP	
SRTP Preference	RTP_Only
Authentication Tag Length	80
MKI on Transmit Stream	Yes
Key Derivation Enable	Yes
Key Derivation Rate	16
Window Size Hint	64
UnEncrypted SRTP Enable	No
UnEncrypted SRTCP Enable	No
UnAuthenticated SRTP Enable	No

3.4.1 Network Group

The Network subgroup includes the following parameters:

- Network Group Label

3.4.1.1 Network Group Label

Description: Label of the currently selected VoIP network group.

Allowed Values: A string with length between 0 - 40 characters

Default Value = Default

3.4.2 Transport Group

The Transport subgroup includes the following parameters:

- Transport Protocol
- SIPS URI Scheme

3.4.2.1 Transport Protocol

Description: Defines the preferred transport protocol of call signaling packets.

Allowed Values:

- UDP = User Datagram Protocol is used as the transport protocol.
- TCP = Transmission Control Protocol is used as the transport protocol.
- TLS = Transport Layer Security is used as the transport protocol.

Default Value = UDP

3.4.2.2 SIPS URI Scheme

Description: Defines whether the SIPS URI (Uniform Resource Identifier) or SIP URI scheme will be used for generating SIP messages.

Allowed Values:

- Yes = All Request, To, From, and Contact URIs generated by the gateway will use the SIPS URI scheme.
- No = All Request, To, From, and Contact URIs generated by the gateway will use the SIP URI scheme.

Default Value = No

3.4.3 URI Parameters Group

The URI Parameters subgroup includes the following parameters:

- User Phone Parameter
- Local Phone Context
- Remote Phone Context

3.4.3.1 User Phone Parameter

Description: Determines if the “user=phone” attribute is included in the SIP URI.

Allowed Values:

- Yes
- No

Default Value = Yes

3.4.3.2 Local Phone Context

Description: Defines the string that will be used in the “phone-context” attribute included in “From” header of the SIP URI.

Allowed Values: String with length between 0 - 128 characters

Default Value = (no default value)

Note: If blank, the phone-context attribute will not be included in the “From” header.

3.4.3.3 Remote Phone Context

Description: Defines the string that will be used in the “phone-context” attribute included in “To” header of the SIP URI.

Allowed Values: String with length between 0 - 128 characters

Default Value = (no default value)

Note: If blank, the phone-context attribute will not be included in the “To” header.

3.4.4 Proxy Group

The Proxy subgroup includes the following parameters:

- Primary Proxy Server Address
- Primary Proxy Server Port
- Backup Proxy Server Address
- Backup Proxy Server Port
- Proxy Query Interval

3.4.4.1 Primary Proxy Server Address

Description: The IP Address of the SIP Proxy Server through which the Media Gateway may send/receive requests. If blank, the Media Gateway will not use a Proxy Server.

Allowed Values: Any valid IP Address in dotted decimal notation

Default Value = (no default value)

3.4.4.2 Primary Proxy Server Port

Description: The IP Port of the SIP Proxy Server. If an IP Address was entered for the Primary Proxy Server Address parameter, then the Primary Proxy Server Port parameter must be set to a valid port number.

Allowed Values: 1024-65000

Default Value = 5060

3.4.4.3 Backup Proxy Server Address

Description: The IP Address of the SIP Backup Proxy Server through which the Media Gateway may send/receive requests. If blank, the Media Gateway will not use a Backup Proxy Server.

Allowed Values: Any valid IP Address in dotted decimal notation.

Default Value = (no default value)

3.4.4.4 Backup Proxy Server Port

Description: The IP Port of the SIP Backup Proxy Server. If an IP Address was entered for the Backup Proxy Server Address parameter, then the Backup Proxy Server Port parameter must be set to a valid port number.

Allowed Values: 1024-65000

Default Value = 5060

3.4.4.5 Proxy Query Interval

Description: Interval in seconds at which the Primary Proxy Server is queried. If the Primary Proxy Server does not respond to the query, the Media Gateway will switch to the Backup Proxy Server. Once the Primary Proxy Server responds to the query, the Media Gateway will switch back to the Primary Proxy Server.

Note: This parameter is only valid if the Backup Proxy Server Address parameter is configured. The Primary Proxy Server must respond to SIP OPTIONS requests in order for the Proxy Query to succeed.

Allowed Values: 10-3600 seconds

Default Value = 30 seconds

3.4.5 Registration Group

The Registration subgroup includes the following parameters:

- Registration Server Address
- Registration Server Port
- Registration User
- Registration Expiration

3.4.5.1 Registration Server Address

Description: IP Address of the SIP Registration Server that the Media Gateway should register with. If blank, the Media Gateway will not register with a Registration Server.

Allowed Values:

- Any valid IP address in dotted decimal notation.
- Blank

Default Value = (no default value)

3.4.5.2 Registration Server Port

Description: IP Port of the SIP Registration Server. If a SIP Registration Server IP Address was entered for the Registration Server Address parameter, then the Registration Server Port parameter must be set to a valid port number.

Allowed Values: 1024 - 65000

Default Value = 5060

INI File Parameter Name = sipRegPort

3.4.5.3 Registration User

Description: Specifies the address-of-record (AOR) to be registered. If it does not contain '@domain' then the gateway IP address is appended.

Allowed Values: A string with length between 0 - 64 characters

Default Value = (no default value)

3.4.5.4 Registration Expiration

Description: Specifies the amount of time in seconds that the registration with the SIP Proxy Server is valid.

Allowed Values: 1 - 60000 seconds

Default Value = 3600 seconds

3.4.6 Audio Group

The Audio subgroup includes the following parameters:

- Codec/Frame Size/Frames Per Packet
- Audio Compression

3.4.6.1 Codec/Frame Size/Frames Per Packet

Description: Refers to the type of coder/decoder used for voice compression. The parameter values for each Codec type are defined in Table 1, "Coder/Decoder Parameters", on page 86.

Note: (For DMG1000 only) If secure RTP (SRTP) is enabled, then you must select 30 milliseconds frame sizes for G.711.

Table 1. Coder/Decoder Parameters

Codec Type	Frame Size in Bytes	Frames Per Packet
DMG1000		
G.711	Selectable: 10, 20, 30	1
G.723	10	Selectable: 1 through 4
G.729	20	Selectable: 1 through 8
DMG2000		
G.711	Selectable: 10, 20, 30	1
G.723	10	1 or 2
G.729	20	Selectable: 1 through 6

INI File Parameter Name = dspFrameSizeG711, dspFramesPerPktG723k, dspFramesPerPktG729

3.4.6.2 Audio Compression

Description: Sets the audio coder/decoder to be used by the Media Gateway.

Allowed Values:

- G.711u/G.711a = G.711 uLaw preferred, G.711 aLaw secondary
- G.711u = G.711 uLaw only
- G.711a = G.711 aLaw only
- G.723.1 = G.723.1 only
- G.729AB = G.729AB only

Default Value = G.711u/G.711a

INI File Parameter Name = dspCompression

Note: Unit requires a restart if this parameter value is changed.

3.4.7 SRTP Group

The Secure Real-time Transport Protocol (RTP) subgroup includes the following parameters:

- SRTP Preference
- MKI on Transmit Stream
- Key Derivation Enable
- Key Derivation Rate
- Anti-replay Window Size Hint
- Cipher Mode
- Authentication Type
- Authentication Tag Length
- Unauthenticated SRTP Enable
- UnEncrypted SRTCP Enable
- UnEncrypted SRTP Enable

SRTP Preference

Description: Specifies whether the Media Gateway and server exchange voice packets as encrypted or non-encrypted.

Allowed Values:

- RTP_Only = The Media Gateway uses only RTP. If the server only supports Secure RTP, then the calls are rejected.
- SRTP_Only = The Media Gateway uses only Secure RTP. If the server does not support Secure RTP, then the calls are rejected.
- SRTP_Preferred = The Media Gateway attempts to connect using Secure RTP. If the server supports Secure RTP, the call is connected using Secure RTP. If the server does not support Secure RTP, then the call is connected using RTP.

  Note: SRTP_Preferred is not supported in Version 5.1 SU2 Software.

Default Value = RTP_Only

INI File Parameter Name = srtpPreference

Note: Unit requires a restart if this parameter value is changed.

MKI on Transmit Stream

Description: Specifies if Master Key Index (MKI) is supported on transmit stream.

Allowed Values:

- Yes = Transmit stream supports MKI
- No = Transmit stream does not support MKI

Default Value = Yes

INI File Parameter Name = srtpTxMkiEnable

Note: Currently, the Media Gateway uses only one master key.

Key Derivation Enable

Description: Specifies if the Secure RTP (SRTP) Key is changed during a voice session.

Allowed Values:

- Yes = The SRTP Key is periodically changed during a voice session. The period is determined by the Key Derivation Rate.
- No = A single SRTP Key is used during a voice session.

Default Value = Yes

INI File Parameter Name = srtpKdrEnable

Key Derivation Rate

Description: Specifies the number of voice packets that cause the SRTP Key to be changed. The number is 2**<KDR>.

Allowed Values: 16 to 24

Default Value = 16

INI File Parameter Name = srtpKdrValue

Anti-replay Window Size Hint

Description: Specifies the Anti-replay window size hint.

Allowed Values: 64 to 99

Default Value = 64

INI File Parameter Name = srtpWsh

Cipher Mode

Description: Specifies the cipher used to encrypt voice packets.

Note: Plain text is essentially no encryption and should be used only for testing purposes. For no encryption, you should specify RTP ONLY in the SRTP Preference parameter.

Allowed Values:

- Plain_Text = Use Plain_Text as the cipher (No encryption - for testing purposes only)
  Note: Plain_Text is not supported in Version 5.1 SU2 Software.
- AES_Counter_Mode = Use AES_Counter_Mode as the cipher

Default Value = AES_Counter_Mode

INI File Parameter Name = srtpEncodeType

Authentication Type

Description: Specifies the type of packet authentication used with Secure RTP. It is recommended that you enable authentication when using Secure RTP.

Allowed Values:

- None = No authentication is used
  Note: None is not supported in Version 5.1 SU2 Software.
- SHA1 = SHA1 authentication type is used

Default Value = SHA1

INI File Parameter Name = srtpAuthType

Authentication Tag Length

Description: Specifies the length of the authentication tag transmitted with the voice packet. A 32-bit tag length should be used when network loading is a concern.

Allowed Values:

- SHA1_32_bit = Uses 32-bit authentication tag
- SHA1_80_bit = Uses 80-bit authentication tag

Default Value = SHA1_80_bit

INI File Parameter Name = srtpAuthTag

Unauthenticated SRTP Enable

Description: This parameter can be used to force Realtime Transport Protocol (RTP) packets to be received without authentication even if an authentication algorithm has been negotiated between parties.

Allowed Values:

- Yes = RTP VOICE packets will NOT be authenticated.
- No = RTP VOICE packets will be authenticated per the negotiated algorithm.

Default Value = No

INI File Parameter Name = srtpUnAuthenticatedSRTP

UnEncrypted SRTCP Enable

Description: This parameter can be used to force SRTCP packets to be transmitted in an unencrypted fashion even if cipher keys have been negotiated between parties.

Allowed Values:

- Yes = Transmitted RTCP CONTROL packets will NOT be encrypted despite the negotiation of cipher keys.
- No = Transmitted RTCP CONTROL packets will be encrypted per the negotiated cipher keys.

Default Value = No

INI File Parameter Name = srtpUnEncryptedSRTCP

UnEncrypted SRTP Enable

Description: This parameter can be used to force Secure Realtime Transport Protocol (SRTP) packets to be transmitted in an unencrypted fashion even if cipher keys have been negotiated between parties.

Allowed Values:

- Yes = Transmitted RTP VOICE packets will NOT be encrypted despite the negotiation of cipher keys.
- No = Transmitted RTP VOICE will be encrypted per the negotiated cipher keys.

Default Value = No

INI File Parameter Name = srtpUnEncryptedSRTP

3.5 VoIP Media Parameters

The VoIP Media parameters include the following subgroups:

- Early Media Group
- Audio Group
- Fax Group
- RTP Group

3.5.1 Early Media Group

The Early Media subgroup includes the following parameters:

- RFC 3960 Early Media Support
- Send Early 183 Progress Response
- Early Media Requires Reliable Provisional Responses

RFC 3960 Early Media Support

Description: Specifies the Early Media mode to be supported by the gateway.

Allowed Values:

- None = Early Media support is disabled.
- Always = Early Media is enabled for all calls (RFC-3960 Gateway Model).
- OnDemand = Early Media is enabled on a call-by-call basis. INVITEs generated by the gateway will include 'early-session' in the Supported header, while INVITEs received by the gateway must include 'early-session' in Supported header to request Early Media (RFC 3960 Application Model).

Default Value = OnDemand

INI File Parameter Name = sipEarlyMediaSupport

Send Early 183 Progress Response

Description: Specifies when 183 Progress Response is sent from the gateway. If set to Yes, the gateway will send 183 Progress Response prior to establishing the connection with far end. If set to No, the gateway will send183 Progress Response after. Early Media Support must be enabled.

Allowed Values:

- Yes
- No

Default Value = No

INI File Parameter Name = sipSendEarlyProgress

Early Media Requires Reliable Provisional Responses

Description: Specifies that Early-Media is only available if reliable provisional responses (PRACK) are supported or required on the call.

Allowed Values:

- Yes
- No

Default Value = No

INI File Parameter Name = sipEarlyMediaReqRelProvRsp

3.5.2 Audio Group

The Audio subgroup includes the following parameters:

- Signaling Digit Relay Mode
- Voice Activity Detection
- Acceptable Media
- Continue Ringback on CN
- Packet Time for Inbound VoIP
- Digit Relay Mode
- Telephone-Event Payload Type
- G722 Enable
- Unhold on RE-INVITE with No SDP

Signaling Digit Relay Mode

Description: Specifies if out-of-band messages are used to transport DTMF tones between the Media Gateway and VoIP endpoints. Specifically, when set to 'On', the Media Gateway will notify VoIP endpoints of received DTMF digits on the TDM interface via an out-of-band message (e.g. SIP INFO).

Note: The Media Gateway will accept out-of-band DTMF messages from VoIP endpoint to render DTMF tones on the TDM interface regardless of the setting of this parameter.

Allowed Values:

- On
- Off

Default Value = Off

INI File Parameter Name = gwSigDigitRelay

Voice Activity Detection

Description: Enables the use of voice activity detection (VAD) to reduce the amount of audio data traffic. If VAD is enabled, the unit will stop the transmission of RTP audio data when no voice activity is detected on the telephony port. Transmission is continued when voice activity is detected. This drastically reduces the amount of audio traffic on the network.

Allowed Values:

- On = Voice activity detection is enabled.
- Off = Voice activity detection is disabled.

Default Value = On

INI File Parameter Name = dspVAD

Acceptable Media

Description: Specifies the Acceptable Media mode to be supported by the gateway. SRTP SDP Offer must contain a=crypto line and can specify m=line with RTP/AVP or RTP/SAVP.

Allowed Values:

- RTP Only = Reject SDP Offer with 488 Not Acceptable Here if it does not specify m=line with RTP/AVP.
- SRTP_Only = Reject SDP Offer with 488 Not Acceptable Here if it does not contain SRTP request.
- RTP_SRTP = Accept RTP and/or SRTP SDP Offers. The gateway will select first m=line specified.

Default Value = RTP_SRTP

INI File Parameter Name = sipAcceptableMedia

Continue Ringback on CN

Description: In early media, ringback will stop when any Comfort Noise or RTP stream is received. This specifies whether to ignore Comfort Noise on early media. If Yes, then on detecting Comfort Noise, ringback will continue to play. If No, then on detecting Comfort Noise, ringback will stop playing. Detecting audio packet in the RTP stream will always stop ringback.

Allowed Values:

- Yes = Ringback will continue to play.
- No = Ringback will stop playing.

Default Value = Yes

INI File Parameter Name = gwIgnoreCn

Packet Time for Inbound VoIP

Description: Specifies a packet time to use when there is no packet time specified by the SDP of Inbound VoIP calls. The actual packet time used may be different depending on negotiated packet time, codec, or system limitations.

Allowed Values:

- 10
- 20
- 30
- 40
- 50
- 60

Codec Type	Valid Packet Times (ms)
G.711	10, 20, 30, 40*, 50*, 60*
G.723	30, 60
G.729	10, 20, 30, 40, 50, 60
G.722	20
* = available on DMG2000 only.	

Note: For DMG1000 only, an orange background color indicates that the current selection (in combination with other data) will limit the available ports to 4.

Default Value = 20

INI File Parameter Name = sipGlobalPacketTime

Digit Relay Mode

Description: Selects the Realtime Transfer Protocol (RTP) method by which dual tone multi-frequency (DTMF) tones are transported between the Media Gateway and VoIP endpoints.

Note: The Inband-Tone method passes the digits as audio data in the RTP stream, and is only reliable when the parameter **Audio Compression** is set to **G.711 only**. The AVT method uses RFC 2833 RTP packets to pass the digits.

Allowed Values (Phone Emulating):

- None = The Media Gateway will not use RTP packets to transmit/receive tone information.
- RFC2833 = The (DTMF) tone information is sent between the Media Gateway and VoIP endpoints via RTP packets as defined by RFC 2833.

  Note: A payload type of 101 is used in the implementation of the RFC2833 method.
- RFC2833-LowLatency = Same as RFC2833, but with 20 milliseconds DTMF detection delay removed. Selecting RFC2833-LowLatency may result in up to 20 milliseconds of DTMF being leaked inband. This also applies to SIP-SIP calls that require transcoding.

  Note: This is available on DMG2000 only.
- Inband-Tone = The DTMF tones are coded into the regular audio packets (RTP voice packets) sent between the Media Gateway and VoIP endpoints.

Default Value = RFC2833

INI File Parameter Name = dspDigitRelay

Telephone-Event Payload Type

Description: Sets the dynamic Payload Type (PT) of the RTP header for telephone-event subtypes. For an inbound call, if a PT is not specified, this is used as the default. This PT will always be specified for the outbound call.

Note: Digit Relay Mode must be set to use RFC2833.

Allowed Values: 96 to 127

Default Value = 101

INI File Parameter Name = sipTelEventPayload

G722 Enable

Description: Enables support for G722 codec. If set to No, then G722 support is disabled. If set to Yes, then G722 support is enabled.

Allowed Values:

- Yes = G722 support is enabled.
- No = G722 support is disabled.

Default Value = No

INI File Parameter Name = sipG722Enabled

Note: Unit requires a restart if this parameter value is changed.

Unhold on RE-INVITE with No SDP

Description: If set to No and the call is on hold, when receiving a RE-INVITE with no SDP, the hold is not removed. If set to Yes and the call is on hold, when receiving a RE-INVITE with no SDP, the hold is removed.

Allowed Values:

- Yes
- No

Default Value = No

INI File Parameter Name = sipReInviteNoSdpIsUnhold

3.5.3 Fax Group

The Fax subgroup includes the following parameters:

- Fax-IP Transport Mode
- Fax Server Address
- Fax Server Network Group
- Fax/Modem Tone Relay Mode
- Enable V.34 Fax
- V.34 Fax Speed Limit

Fax-IP Transport Mode

Description: Defines the method used by the Media Gateway to transport fax calls over IP.

Allowed Values:

- None = No special processing is performed for transporting fax.
- T.38 = T.38 fax protocol is used.
- G.711- Passthrough = Media Gateway will automatically switch the IP media stream to G.711 Passthrough.

Default Value = T.38

INI File Parameter Name = gwFaxTransportMode

Fax Server Address

Description: Defines the IP address to which all inbound TDM fax calls will be directed when non-null.

Allowed Values: IP address in dotted decimal notation

Default Value = (no default value)

INI File Parameter Name = gwFaxServerAddress

Fax Server Network Group

Description: Defines the Network Group to be used for inbound TDM fax calls when Fax Server Address is non-null.

Allowed Values: Any string with length between 0 - 63 characters

Default Value = (no default value)

INI File Parameter Name = gwFaxServerNetworkGroup

Fax/Modem Tone Relay Mode

Description: Selects the Realtime Transfer Protocol (RTP) method by which fax and modem tones are transported between the TDM network and VoIP network.

Allowed Values:

- RFC2833 = Fax and modem tones from the TDM network are sent as RFC2833 packets to the VoIP network. RFC2833 packets from the VoIP network are rendered to TDM.
- Inband-Tone = Fax and modem tones from the TDM network are passed through VoIP. Inband-Tone is only reliable when the Preferred Code parameter is set to G.711 only.

Default Value = RFC2833

INI File Parameter Name = dspFaxModemToneRelay

Enable V.34 Fax

Description: The gateway supports V.17 and V.34 fax modems.

Note: The V.34 fax modem setting is currently only supported on the DMG2030DTIQ and DMG2060DTISQV34.

Allowed Values:

- Yes = The gateway will use V.34 fax modem.
- No = The gateway will not use V.34 fax modem but instead will use V.17 fax modem.

Default Value = No

INI File Parameter Name = gwV34Enable

V.34 Fax Speed Limit

Description: The gateway V.34 fax modem supports rate configurability.

Note: The V.34 fax modem rate setting is currently only supported on the DMG2030DTIQ and DMG2060DTISQV34.

Allowed Values: {33600, 31200, 28800, 26400, 24000, 21600, 19200, 16800, 14400, 12000, 9600, 7200, 4800, 2400, None}

Default Value = 33600

INI File Parameter Name = gwV34SpeedLimit

3.5.4 RTP Group

The Real-time Transport Protocol (RTP) subgroup includes the following parameters:

- RTP Start Port
- RTP End Port
- RTP Source IP Address Validation
- RTP Source UDP Port Validation
- RTP QoS Byte

RTP Start Port

Description: Specifies the first UDP IP port used for Realtime Transport Protocol (RTP) traffic. RTP port number assignments increment from this starting value.

Allowed Values: 1024 to 65000

Default Value = 49000

INI File Parameter Name = gwRTPStartPort

Note: Unit requires a restart if this parameter value is changed.

RTP End Port

Description: Specifies the last UDP IP port used for Realtime Transport Protocol (RTP) traffic. This value must be greater than the Start Port for RTP. It is recommended that this value be at least 1000 more than Start Port for RTP.

Allowed Values: 1024 to 65000

Default Value = 50000

INI File Parameter Name = gwRTPEndPort

Note: Unit requires a restart if this parameter value is changed.

RTP Source IP Address Validation

Description: If set to On, then the source IP address of received RTP packets must match the IP address to which RTP is being sent. If the source IP address does not match, then the packet is discarded. If set to Off, then the source IP address of received RTP packets is not validated.

Allowed Values:

- On
- Off

Default Value = Off

INI File Parameter Name = gwRTPValidateSrcIp

Note: Unit requires a restart if this parameter value is changed.

RTP Source UDP Port Validation

Description: If set to On, then the source UDP port of received RTP packets must match the UDP port to which RTP is being sent. If the source UDP port does not match, then the packet is discarded. If set to Off, then the source UDP port of received RTP packets is not validated.

Allowed Values:

- On
- Off

Default Value = Off

INI File Parameter Name = gwRTPValidateSrcPort

Note: Unit requires a restart if this parameter value is changed.

RTP QoS Byte

Description: The RTP QoS Type parameter defines a decimal value that represents QoS bit flags. This parameter is used in each RTP data packet transmitted from the Media Gateway with a QoS byte code. Routers use this byte code to assign priority levels to packets. The QoS byte may be interpreted as either IPv4 TOS or DiffServ.

IPv4 TOS Byte

Bits:
0-2 = Precedence (RFC 1122)
3-6 = Type of Service (RFC 1349/1455)
7 = Must be zero

PrecedenceBits (0-2):
111xxxxx (224) = Network Control
110xxxxx (192) = Internetwork Control
101xxxxx (160) = CRITIC/ECP
100xxxxx (128) = Flash Override
011xxxxx (96) = Flash
010xxxxx (64) = Immediate
001xxxxx (32 = Priority
000xxxxx (0) = Routine

Type of Service Bits (3-6):
xxx0000x (0) = Normal
xxx1000x (16) = Minimize delay
xxx0100x (8) = Maximize throughput
xxx0010x (4) = Maximize reliability
xxx0001x (2) = Minimize monetary cost
xxx1111x (30) = Maximize physical link security
xxx (32 = Priority
xxx (0) = Routine

Default Value = 0 (Routine/Normal)

DiffServ Codepoint Byte

Bits:
0-5 = Differentiated Services Codepoint (RFC 2474)
6-7 = Explicit Congestion Notification (RFC 2481)

Allowed Values: 0-255

Default Value = 0

INI File Parameter Name = gwQosRtp

3.6 VoIP Authentication Parameters

Information about VoIP Authentication and how it is supported by the Dialogic® Media Gateway is described in the following sections:

- VoIP Authentication Overview
- Inbound VoIP (Server)
- Outbound VoIP (Client)

3.6.1 VoIP Authentication Overview

The VoIP Authentication feature on the gateway provides a means for validating SIP devices requesting services of the gateway (Inbound VoIP) and for the gateway to respond to authentication requests when the gateway is requesting services from a SIP device (Outbound VoIP). To use this feature, navigate to the *Configuration-->VoIP-->Authentication* link via the Media Gateway's user interface.

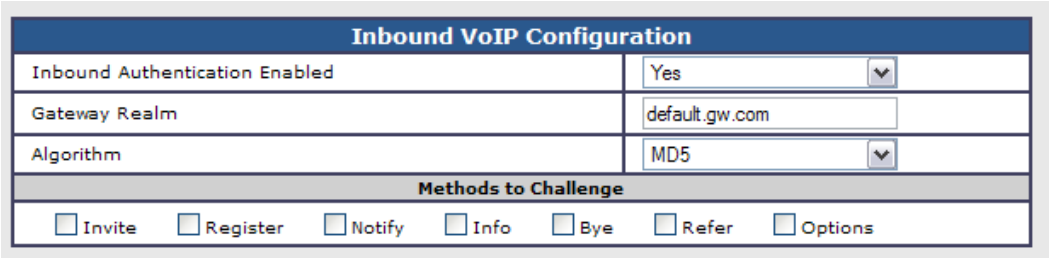
3.6.1.1 Inbound VoIP (Server)

Inbound VoIP refers to the gateway acting as a SIP server and receiving SIP methods. Inbound VoIP may be used to prevent unauthenticated SIP agents from routing calls through the gateway.

3.6.1.2 Inbound VoIP Configuration

Figure 13 is a sample screen shot of the Inbound VoIP Configuration Web page:

Figure 13. Inbound VoIP Configuration Web Page



Inbound Authentication Enabled - Determines if the gateway will challenge a peer upon receiving a SIP method. If Yes and the appropriate Method to Challenge is selected through the checkbox, the gateway will challenge the peer upon reception of the method by sending a SIP 401 response along with the Gateway Realm.

Gateway Realm - Defines a protection space of the gateway. Associated with the Gateway Realm is a series of User Name and Password pairings. The gateway may challenge a peer with the Realm. The peer must have knowledge of the User Name and Password pairing in order to be authenticated by the gateway.

Algorithm - Sets the algorithm of the gateway. MD5 is the default hashing algorithm for authentication as specified by RFC 2617. MD5-sess is the algorithm that employs MD5 hashing with the use of a session key.

Methods to Challenge - Determines which received SIP methods will be challenged if the Inbound Authentication Enabled is set to Yes.

- Invite
- Register
- Notify
- Info
- Bye
- Refer
- Options

3.6.1.3 Users

Figure 14 is a sample screen shot of the Users Web page:

Figure 14. Users Web Page

Realm - Specifies the Gateway Realm listed in the Inbound VoIP Configuration section.

User Name - Specifies the User Name identifier associated with the Gateway Realm.

Password - Specifies the Password associated with the User Name.

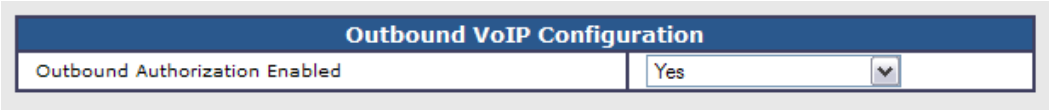
3.6.2 Outbound VoIP (Client)

Outbound VoIP refers to the gateway acting as a client and sending SIP methods (INVITE, REGISTER, NOTIFY, etc.) to a SIP peer acting as a server. Outbound VoIP could be used to verify the gateway to a SIP proxy server or other SIP agents.

3.6.2.1 Outbound VoIP Configuration

Figure 15 is a sample screen shot of the Outbound VoIP Configuration Web page:

Figure 15. Outbound VoIP Configuration Web Page

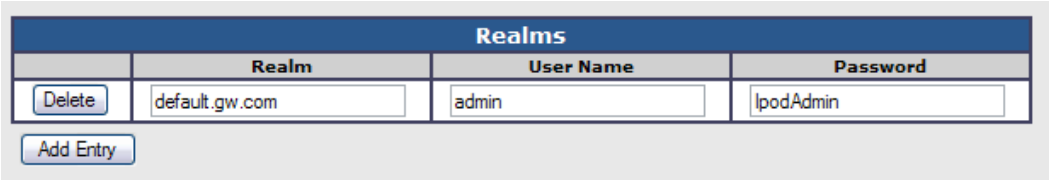


Outbound Authorization Enabled - Enables or disables the ability of the gateway to respond to Authorization challenges. If Yes, the gateway is capable of resending a SIP method with Authorization information as a result of a received challenge from a SIP peer.

3.6.2.2 Realms

Figure 16 is a sample screen shot of the Realms Web page:

Figure 16. Realms Web Page



Realm - Specifies the Realm protection of another SIP peer. Realm is the User Name and Password pairing. When challenged by a SIP peer with Realm, the gateway must have the User Name and Password in the SIP peer's protection space.

User Name - Specifies the User Name identifier associated with the Realm of the SIP peer.

Password - Specifies the Password associated with the User Name.

3.7 TDM General Parameters

The TDM General Settings group includes the following parameters:

- PCM Coding
- Minimum Call Party Delay (Phone Emulating Only)
- Maximum Call Party Delay (Phone Emulating Only)
- Dial Digit On Time
- Dial Inter-Digit Time
- Dial Pause Time
- Turn MWI On FAC (Phone Emulating Only)
- Turn MWI Off FAC (Phone Emulating Only)
- Outbound Call Connect Timeout (Phone Emulating Only)
- Wait for Ringback/Connect on Blind Transfer (Phone Emulating Only)
- Hunt Group Extension (Phone Emulating Only)
- Disconnect on Fax Cleardown Tone
- Connect Outbound Call On DTMF

PCM Coding

Description: Sets the Media Gateway to the PCM coding mode used by the PBX.

Allowed Values:

- µLaw = PBX uses µLaw coding.
- aLaw = PBX uses ALaw coding.

Default Value = µLaw

INI File Parameter Name = dspPcmCoding

Minimum Call Party Delay (Phone Emulating Only)

Description: Specifies the minimum number of milliseconds that the Media Gateway will wait for display information (calling party information) on an inbound PBX call before the call is routed to the IP destination. (0 for no delay)

Allowed Values: 0 through 10000 milliseconds

Default Value = 0

INI File Parameter Name = telMinWaitForCpid

Maximum Call Party Delay (Phone Emulating Only)

Description: Specifies the maximum number of milliseconds that the Media Gateway will wait for display information (calling party information) on an inbound PBX call before the call is routed to the IP destination. (0 for no delay)

Allowed Values: 0 through 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telCallInfoTout

Dial Digit On Time

Description: Specifies the duration in milliseconds of a dialed DTMF digit to the PBX.

Allowed Values: 30 through 2000 milliseconds

Default Value = 100 milliseconds

INI File Parameter Name = telDialDigitOnMs

Dial Inter-Digit Time

Description: Specifies the delay in milliseconds between dialed DTMF digits to the PBX.

Allowed Values: 30 through 2000 milliseconds

Default Value = 100 milliseconds

INI File Parameter Name = telDialInterDigitMs

Dial Pause Time

Description: Specifies the delay in milliseconds for each pause (,) character encountered in a dial string.

Allowed Values: 40 through 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telDialPauseMs

Turn MWI On FAC (Phone Emulating Only)

Description: Specifies the Features Access Code (FAC) to dial to turn on the Message Waiting Indicator (MWI) of a PBX extension.

Allowed Values: Any string of less than 11 characters

Default Value = (no default value)

INI File Parameter Name = telMwiOnFAC

Turn MWI Off FAC (Phone Emulating Only)

Description: Specifies the Features Access Code (FAC) to dial to turn off the Message Waiting Indicator (MWI) of a PBX extension.

Allowed Values: Any string of less than 11 characters

Default Value = (no default value)

INI File Parameter Name = telMwiOffFAC

Outbound Call Connect Timeout (Phone Emulating Only)

Description: Specifies the time in milliseconds that the Media Gateway will wait for a connect event on an outbound circuit call. If no connect event is received within the time specified, the Media Gateway will automatically transition the call to the connected state.

Allowed Values: 0 to 60000 milliseconds

Default Value = 10000 milliseconds

INI File Parameter Name = telConnectToutMs

Wait for Ringback/Connect on Blind Transfer (Phone Emulating Only)

Description: Specifies whether the gateway will wait for the detection of ringback tone or connection before completing a blind transfer of a circuit call. If 'Yes' is selected, blind transfers will wait for a ringback tone or connect event (voice) before completing the transfer. If 'No' is selected, a blind transfer will be completed as soon as the destination number is dialed. Valid only in OnAnswer connect mode.

Allowed Values: Yes, No

Default Value = Yes

INI File Parameter Name = telBlindXfrWaitConfirm

Hunt Group Extension (Phone Emulating Only)

Description: If the telephony ports are configured in a Hunt Group on the switch, then enter the Hunt Group extension number in this field. The gateway will use this field to ignore the Hunt Group extension when generating the call party information. This ensures that the Hunt Group extension number is not mistaken for a calling or called party number.

Allowed Values: Any valid dialable number string up to 7 characters

Default Value = (no default value)

INI File Parameter Name = telHuntGroupExtn

Disconnect on Fax Cleardown Tone

Description: Specifies whether the gateway will disconnect the call when fax cleardown tone is received.

Allowed Values:

- Yes = The call will be disconnected when the fax is complete.
- No = The call will not be disconnected when the fax is complete.

Default Value = No

INI File Parameter Name = telDisconnectOnFaxCleardown

Connect Outbound Call On DTMF

Description: Specifies whether the gateway will connect the outbound call when it is in the Alerting or Proceeding state and a DTMF event is detected.

Allowed Values:

- Yes = An outbound call is in the Alerting or Proceeding state and a DTMF event is detected, the call will transition to the Connected state.
- No = An outbound call is in the Alerting or Proceeding state and a DTMF event is detected, the event will be discarded and the call state will remain unchanged.

Default Value = No

INI File Parameter Name = telConnectOnDTMFEnable

3.8 TDM T1/E1 Parameters

The T1/E1 parameters are used to configure the T1 and E1 port(s) on DMG2000 only and include the following groups:

- T1/E1 Mode Group
- T1 CAS Protocol Group (T1 CAS Signaling Mode)
- T1 ISDN Protocol Group (ISDN Signaling Mode)
- E1 ISDN Protocol Group (ISDN Signaling Mode)

3.8.1 T1/E1 Mode Group

The parameters in the T1/E1 Mode group include the following:

- Line Mode
- Signaling Mode
- Telephony Port Interface Side
- Clock Control

3.8.1.1 Line Mode

Description: Specifies the Line Mode type (T1 or E1) to which the T1/E1 connector will be interfacing.

Allowed Values:

- T1 = Interfaces to T1 line (23/24 channels)
- E1 = Interfaces to E1 line (30 channels)

Default Value = T1

INI File Parameter Name = t1e1LineMode

3.8.1.2 Signaling Mode

Description: Specifies the signaling mode to be used.

Allowed Values:

- CAS = Channel associated signaling
- ISDN = Common channel signaling

Default Value = CAS

INI File Parameter Name = t1e1Signaling

Note: Unit requires a restart if this parameter value is changed.

3.8.1.3 Telephony Port Interface Side

Description: Identifies the side of the connection.

Allowed Values:

- Terminal = Port is Terminal Equipment (TE)
- Network = Port is Network Termination (NT)

Default Value = Terminal

INI File Parameter Name = telPortInterfaceSide

3.8.1.4 Clock Control

Description: Selects if a port receives clock from external equipment or drives clock to external equipment.

Allowed Values:

- Automatic
- Receive
- Drive

Default Value = Automatic

INI File Parameter Name = telPortTxClkCreate

3.8.2 T1 CAS Protocol Group (T1 CAS Signaling Mode)

The parameters in the T1 CAS Protocol group include the following:

- T1 CAS Protocol
- Flash Hook
- Wait for Dial Tone after Flash Hook
- Delay After Flash-Hook
- Incoming Rings Before Answer
- Ringing Timeout
- Ring Cycle Time
- Enable Glare Detection
- Transfer Feature Code
- Consult Call Dialtone Drop Code
- Consult Call Proceeding Drop Code
- Consult Call Busy Drop Code
- Consult Call Connected Drop Code

- Consult Call Disconnected Drop Code
- Consult Call Error Drop Code
- MWI Confirmation Tone
- Use Same Port for MWI Clear/Set
- Initial Wait for Inband CPID
- Inband CPID Complete Timeout
- Inband Type I CID to First Ring Timeout

3.8.2.1 T1 CAS Protocol

Description: If T1 CAS is selected as the Signaling Mode, specifies the T1 CAS protocol to be used.

Allowed Values:

- Loop Start = Loop Start protocol will be used.
- Ground Start = Ground Start protocol will be used.
- E&M Immediate = E&M Immediate protocol will be used.
- E&M Delay = E&M Delay protocol will be used.
- E&M Wink = E&M Wink protocol will be used.

Default Value = Loop Start

INI File Parameter Name = t1e1CASProtocol

3.8.2.2 Flash Hook

Description: - Specifies the duration in milliseconds that the Media Gateway will remain on-hook during a hook flash operation.

Allowed Values: 50 to 4000 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = telFlashMs

3.8.2.3 Wait for Dial Tone after Flash Hook

Description: Indicate if the gateway should wait for dial tone after flash hook.

Allowed Values:

- Yes = Gateway will wait for dial tone after a Flash Hook as a confirmation that the switch recognized the event.
- No = Gateway does not expect the switch to create dial tone after a Flash Hook.

Default Value = Yes

INI File Parameter Name = telDialToneExpAfterFlashHook

3.8.2.4 Delay After Flash-Hook

Description: Specifies the duration (in milliseconds) to delay after a Flash Hook if a dial tone confirmation is not expected.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 100 to 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telFlashHookNoDialToneDelayMs

3.8.2.5 Incoming Rings Before Answer

Description: - Specifies the number of ring bursts that must be present at the DMG2000 before the call is presented to the IP destination as a new call.

Allowed Values: 1 to 100

Default Value = 1

INI File Parameter Name = telIncomRing

3.8.2.6 Ringing Timeout

Description: - Specifies the number of milliseconds of non-ringing that will signal that an incoming call has gone away.

Allowed Values: 100 to 10000 milliseconds

Default Value = 6000 milliseconds

INI File Parameter Name = telRingOffMs

3.8.2.7 Ring Cycle Time

Description: Specifies the number of milliseconds of non-ringing that will signal that an incoming call has gone away.

Allowed Values: Numerical value from 1000 to 10000

Default Value = 6000

INI File Parameter Name = msRingCycleTime

3.8.2.8 Enable Glare Detection

Description: When making an outbound call on the TDM network, it specifies whether the gateway will treat the failure to detect Dialtone as a Glare Condition. A Glare condition is when an incoming call arrives at the same time a port goes off hook to dial.

Allowed Values:

- Yes = The failure to detect dialtone will be treated as a Glare.
- No = The outbound call will proceed.

Default Value = Yes

INI File Parameter Name = telGlareDetectEnable

3.8.2.9 Transfer Feature Code

Description: Defines the Feature Code to dial in order initiate a transfer.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacTransfer

3.8.2.10 Consult Call Dialtone Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Dialtone state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropDt

3.8.2.11 Consult Call Proceeding Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Proceeding state (dialed but not connected) and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropProc

3.8.2.12 Consult Call Busy Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Busy state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropBusy

3.8.2.13 Consult Call Connected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Connected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropCon

3.8.2.14 Consult Call Disconnected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Disconnected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropDis

3.8.2.15 Consult Call Error Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Error state and to reconnect to the original call.

Allowed Values: Dialable number, including '!' for flash hook and 'p' for pause.

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropError

3.8.2.16 MWI Confirmation Tone

Description: Specifies whether or not the PBX sends a confirmation tone to signal the successful completion of a Message Waiting Indication (MWI) request.

Allowed Values:

- Yes = PBX will send a confirmation tone.
- No = PBX will not send a confirmation tone.

Default Value = No

INI File Parameter Name = telMwiConfirm

3.8.2.17 Use Same Port for MWI Clear/Set

Description: Specifies if the PBX requires that the telephony station that set an MWI be used to clear the MWI.

Note: Some PBX types require that the port that set an MWI be the same port that clears the MWI.

Allowed Values:

- Yes = Use the same port to set and clear the MWI.
- No = Not necessary to use the same port to clear an MWI that was used to set the MWI.

Default Value = Yes

INI File Parameter Name = telMwiSamePort

3.8.2.18 Initial Wait for Inband CPID

Description: Specifies the number of milliseconds the DMG2000 will wait for the first inband DTMF digit to arrive after answering an incoming PBX call.

Allowed Values: 100 to 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telInbCpidStartMs

3.8.2.19 Inband CPID Complete Timeout

Description: Specifies the number of milliseconds the DMG2000 will wait for each subsequent inband DTMF digit to arrive. If no digit is received within this time, the CPID is assumed to be complete.

Allowed Values: 100 to 2000 milliseconds

Default Value = 300 milliseconds

INI File Parameter Name = telInbCpidEndMs

3.8.2.20 Inband Type I CID to First Ring Timeout

Description: Specifies the number of milliseconds that are allowed from the end of the Type I CPID on the first ring. If this time is exceeded, the CPID is cleared.

Note: This parameter is only needed when CPID arrives at the Gateway before the first ring. This parameter can be ignored if the CPID arrives between the first and second ring.

Allowed Values: 500 to 30000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telPreRingDtmfCidTimeoutMs

3.8.3 T1 ISDN Protocol Group (ISDN Signaling Mode)

The parameters in the T1 ISDN Protocol group include the following:

- Line Encoding
- Framing
- Selects Transmit Pulse Waveform
- ISDN Protocol
- ISDN Protocol Variant
- Multiple Diversion Processing
- Network Specific Facilities (NSF)
- ISDN Answer Supervision Enable
- Enable Failover

3.8.3.1 Line Encoding

Description: Specifies the type of T1 line encoding that will be used.

Allowed Values:

- AMI = Alternate Mark Inversion line coding is used.
- B8ZS = Binary Eight Zero Substitution line coding is used.

Default Value = B8ZS

INI File Parameter Name = t1Encoding

3.8.3.2 Framing

Description: Specifies the type of T1 framing that will be used by the line.

Allowed Values:

- SF = D4 Superframe format (12 consecutive T1 frames) is used.
- ESF = Extended Superframe format (24 consecutive T1 frames) is used.

Default Value = ESF

INI File Parameter Name = t1Framing

3.8.3.3 Selects Transmit Pulse Waveform

Description: Selects the method to be used for generating the Transmit Waveform Shape.

Allowed Values:

- Short Haul 110 ft = The DMG2000 is 0 - 110 feet from T1 line source.
- Short Haul 220 ft = The DMG2000 is 110 - 220 feet from T1 line source.
- Short Haul 330 ft = The DMG2000 is 220 - 330 feet from T1 line source.
- Short Haul 440 ft = The DMG2000 is 330 - 440 feet from T1 line source.
- Short Haul 550 ft = The DMG2000 is 440 - 550 feet from T1 line source.
- Short Haul 660 ft = The DMG2000 is 550 - 660 feet from T1 line source.

Default Value = Short Haul 110 ft

INI File Parameter Name = t1TxWave

3.8.3.4 ISDN Protocol

Description: If ISDN is selected as the Signaling Mode, specifies the T1 ISDN protocol to be used.

Allowed Values:

- QSIG = QSIG ISDN protocol is used.
- NI-2 = US National ISDN - Phase 2 protocol is used.
- 5ESS = 5ESS protocol is used.
- DMS100 = DMS100 protocol is used.

Default Value = QSIG

INI File Parameter Name = t1IsdnProtocol

3.8.3.5 ISDN Protocol Variant

Description: Specifies the type of T1 ISDN protocol variant to be used.

Allowed Values:

- None = None (Standard)
- Alcatel = Support Alcatel Extensions
- Ericsson = Support Ericsson Extensions
- Avaya_IP_Office = Support Avaya IP Office Extensions
- Nortel_DMS-100 = Support Nortel DMS-100 Extensions

Default Value = None

INI File Parameter Name = t1IsdnProtocolVariant

3.8.3.6 Multiple Diversion Processing

Description: Selects which served (diverting) user to process when a call is received with multiple stages of diversion.

User A -> User B1 (fwd) -> User B2 (fwd) -> User C

Notes:

1. For NI-2, Multiple Diversion Processing selects the Redirecting Number (IE 0x74) to use if multiple Redirecting Number IEs are received. A maximum of two Redirecting Numbers is supported.
2. For QSIG, Multiple Diversion Processing selects the served (diverting) user to process when a call is received with multiple stages of diversion.

Allowed Values:

- First = First served user (User B1)
- Second = Second served user (User B2)

Default Value = First

INI File Parameter Name = isdnMultipleDiversion

3.8.3.7 Network Specific Facilities (NSF)

Description: If NI-2 protocol is used and any value other than 'None' is selected, then the specified service type is included as an NSF Information Element in the outgoing ISDN packet.

Allowed Values:

- None = None (Default)
- IntraLATA_OUTWATS = National ISDN OUTWATS Selection
- Foreign_Exchange = Foreign Exchange Selection
- TIE_Trunk = Tie Trunk Selection
- Selective = SCOS (Selective Class of Call Screening) Service Selection
- Access_VPN = Access for Virtual Private Network
- Megacom_800 = MEGACOM 800 telecommunications service
- Megacom = MEGACOM telecommunications service
- Accunet = ACCUNET Switched Digital Services
- Long_Distance = International Long Distance Service
- International_800 = International 800
- Private= Private Virtual Network
- MultiQuest = AT&T DIAL-IT 900 and MultiQuest
- INWATS = National ISDN INWATS
- Hotel_Motel = Hotel/Motel Service Selection

Default Value = None

INI File Parameter Name = t1IsdnNsfIeServices

3.8.3.8 ISDN Answer Supervision Enable

Description: If 'Yes', and an outbound call receives a Progress Indicator with a 1 or 8 (non-ISDN, inband), then the gateway will use in-band answer-supervision to determine when the call has been answered. If 'No', then a received CONNECT message will be used to determine when the call has been answered.

Notes: ***1.*** Progress Indicator 1-Call is not end-to-end ISDN or may be in-band information.

2. Progress Indicator 8-Inband treatment has been applied.

Allowed Values:

- Yes
- No

Default Value = Yes

INI File Parameter Name = isdnAnswerSupervisionEnable

3.8.3.9 Enable Failover

Description: Specifies the Failover mode used when the gateway is powered down.

Note: This parameter is only applicable for DMG2060DTISQ and DMG2120DTISQ models.

Allowed Values:

- Yes = Failover enabled (span 1 to span 2) and (span 3 to span 4).
- No = Failover disabled (spans not interconnected).

Default Value = No

INI File Parameter Name = t1e1Failover

3.8.4 E1 ISDN Protocol Group (ISDN Signaling Mode)

The parameters in the E1 ISDN Protocol group include the following:

- Line Coding
- Framing
- Selects Transmit Pulse Waveform
- ISDN Protocol
- ISDN Protocol Variant
- Contiguous B-Channel
- Multiple Diversion Processing
- Outbound TDM Calling Party Source
- Static TDM Calling Party
- ISDN Answer Supervision Enable
- Enable Failover

3.8.4.1 Line Coding

Description: Specifies the type of E1 line coding that will be used.

Allowed Values:

- AMI = Alternate Mark Inversion line coding is used.
- HDB3 = High Density Bipolar Three line coding is used.

Default Value = HDB3

INI File Parameter Name = e1Encoding

Note: Unit requires a restart if this parameter value is changed.

3.8.4.2 Framing

Description: Specifies the type of E1 framing that will be used by the line.

Allowed Values:

- CRC_MF = Multiframe format with CRC is used.
- FR = Basic frame format is used.
- MF = Multiframe format is used.

Default Value =CRC_FMF

INI File Parameter Name = e1Framing

Note: Unit requires a restart if this parameter value is changed.

3.8.4.3 Selects Transmit Pulse Waveform

Description: Selects the method to be used for generating the Transmit Waveform Shape.

Allowed Values:

- 75_Ohm = 75 ohm (unbalanced) impedance is used.
- 120_Ohm = 120 ohm (balanced) impedance is used.

Default Value = 75 Ohm Impedance

INI File Parameter Name = e1TxWave

Note: Unit requires a restart if this parameter value is changed.

3.8.4.4 ISDN Protocol

Description: If ISDN is selected as the Signaling Mode, specifies the ISDN protocol to be used.

Allowed Values:

- QSIG = QSIG ISDN protocol is used.
- ETSI = EuroISDN (ETSI) protocol is used.

Default Value = QSIG

INI File Parameter Name = e1IsdnProtocol

Note: Unit requires a restart if this parameter value is changed.

3.8.4.5 ISDN Protocol Variant

Description: Specifies the type of E1 ISDN protocol variant to be used.

Allowed Values:

- None = None (Standard)
- Alcatel = Support Alcatel Extensions
- Ericsson = Support Ericsson Extensions
- Avaya IP Office = Support Avaya IP Office Extensions

Default Value = None

INI File Parameter Name = e1IsdnProtocolVariant

3.8.4.6 Contiguous B-Channel

Description: Specifies B-Channel Selection Mode Used By Switch. Only applicable to E1 ISDN protocols.

Allowed Values:

- Yes = Contiguous B-Channel Selection (logical mapping).
- No = Skip B-Channel 16 (physical mapping)

Default Value = No

INI File Parameter Name = e1ContiguousBchan

3.8.4.7 Multiple Diversion Processing

Description: Selects which served (diverting) user to process when a call is received with multiple stages of diversion.

User A -> User B1 (fwd) -> User B2 (fwd) -> User C

Notes: *1.* For NI-2, Multiple Diversion Processing selects the Redirecting Number (IE 0x74) to use if multiple Redirecting Number IEs are received. A maximum of two Redirecting Numbers is supported.

2. For QSIG, Multiple Diversion Processing selects the served (diverting) user to process when a call is received with multiple stages of diversion.

Allowed Values:

- First = First served user (User B1)
- Second = Second served user (User B2)

Default Value = First

INI File Parameter Name = isdnMultipleDiversion

3.8.4.8 Outbound TDM Calling Party Source

Description: The source of the calling party that will be provided to the TDM interface for outbound calls and MWI requests.

Note: When a DMG2000 is connected to a Mitel PBX and is configured for E1 QSIG, this parameter must be set to one of the following options in conjunction with the Static TDM Calling Party parameter:

- Outbound TDM Calling Party Source set to **Static** and Static TDM Calling Party set to a **number** (e.g. 1234).
- Outbound TDM Calling Party Source set to **VoIP** and the VoIP endpoint must provide a calling party number in the VoIP call/MWI call request.
- Outbound TDM Calling Party Source set to **VoIP_Preferred** and Static TDM Calling Party set to a **number** (e.g. 1234).

Allowed Values:

- None = No calling party is sent.
- VoIP = Calling party is derived from the VoIP call/mwi-request.
- Static = A statically configured calling party is used.
- VoIP_Preferred = Calling party is derived from the VoIP call/mwi-request. If not available from VoIP, then the statically configured calling party is used.

Default Value = Static

INI File Parameter Name = gwTdmOutboundCallingPartySrc

3.8.4.9 Static TDM Calling Party

Description: Specifies the calling party that will be provided to the TDM interface if Outbound TDM Calling Party Source is set to Static or VoIP_Preferred.

Note: When a DMG2000 is connected to a Mitel PBX and is configured for E1 QSIG, this parameter must be set to one of the following options in conjunction with the Outbound TDM Calling Party Source parameter:

- Outbound TDM Calling Party Source set to **Static** and Static TDM Calling Party set to a **number** (e.g. 1234).
- Outbound TDM Calling Party Source set to **VoIP** and the VoIP endpoint must provide a calling party number in the VoIP call/MWI call request.
- Outbound TDM Calling Party Source set to **VoIP_Preferred** and Static TDM Calling Party set to a **number** (e.g. 1234).

Allowed Values: Any valid dialable number

Default Value = 1234

INI File Parameter Name = gwTdmOutboundCallingParty

3.8.4.10 ISDN Answer Supervision Enable

Description: If 'Yes', and an outbound call receives a Progress Indicator with a 1 or 8 (non-ISDN, inband), then the gateway will use in-band answer-supervision to determine when the call has been answered. If 'No', then a received CONNECT message will be used to determine when the call has been answered.

Notes: ***1.*** Progress Indicator 1-Call is not end-to-end ISDN or may be in-band information.

2. Progress Indicator 8-Inband treatment has been applied.

Allowed Values:

- Yes
- No

Default Value = Yes

INI File Parameter Name = isdnAnswerSupervisionEnable

3.8.4.11 Enable Failover

Description: Specifies the Failover mode used when the gateway is powered down.

Note: This parameter is only applicable for DMG2060DTISQ and DMG2120DTISQ models.

Allowed Values:

- Yes = Failover enabled (span 1 to span 2) and (span 3 to span 4).
- No = Failover disabled (spans not interconnected).

Default Value = No

INI File Parameter Name = t1e1Failover

3.9 TDM Analog Parameters

The TDM Analog parameters only apply to the Models DMG1008LSW, DMG1004LSW. The Analog parameters include the following subgroups:

- Timing Group
- Feature Code Group
- Message Waiting Control Group
- CPID Settings Group

3.9.1 Timing Group

The Analog parameters in the Timing group include:

- Flash Hook
- Loop Current Off Debounce
- Incoming Rings Before Answer
- Ringing Timeout

3.9.1.1 Flash Hook

Description: Specifies the duration in milliseconds that the DMG1000 will remain on-hook during a hook flash operation.

Allowed Values: 50 to 4000 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = telFlashMs

3.9.1.2 Loop Current Off Debounce

Description: Specifies the time in milliseconds that loop current can be removed by the PBX on an active call before the DMG1000 considers the line disconnected.

Allowed Values: 500 to 10000 milliseconds

Default Value = 2500 milliseconds

INI File Parameter Name = telLCOff

3.9.1.3 Incoming Rings Before Answer

Description: Specifies the number of ring bursts that must be present at the DMG1000 before the call is presented to the IP destination as a new call.

Allowed Values: 1 to 100

Default Value = 1

INI File Parameter Name = telIncomRing

3.9.1.4 Ringing Timeout

Description: Specifies the number of milliseconds of non-ringing that will signal that an incoming call has gone away.

Allowed Values: 100 to 10000 milliseconds

Default Value = 6000 milliseconds

INI File Parameter Name = telRingOffMs

3.9.2 Feature Code Group

The Analog parameters in the Feature Code group include:

- Transfer Feature Code
- Consult Call Dialtone Drop Code
- Consult Call Proceeding Drop Code
- Consult Call Busy Drop Code
- Consult Call Connected Drop Code
- Consult Call Disconnected Drop Code
- Consult Call Error Drop Code

3.9.2.1 Transfer Feature Code

Description: Defines the Feature Code to dial in order initiate a transfer.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacTransfer

3.9.2.2 Consult Call Dialtone Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Dialtone state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropDt

3.9.2.3 Consult Call Proceeding Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Proceeding state (dialed but not connected) and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropProc

3.9.2.4 Consult Call Busy Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Busy state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropBusy

3.9.2.5 Consult Call Connected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Connected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropCon

3.9.2.6 Consult Call Disconnected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Disconnected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropDis

3.9.2.7 Consult Call Error Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Error state and to reconnect to the original call.

Allowed Values: Dialable number, including '!' for flash hook and 'p' for pause.

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropError

3.9.3 Message Waiting Control Group

The Analog parameters in the Message Waiting Control group include:

- MWI Confirmation Tone
- Use Same Port for MWI Clear/Set

3.9.3.1 MWI Confirmation Tone

Description: Specifies whether or not the PBX sends a confirmation tone to signal the successful completion of a Message Waiting Indication (MWI) request.

Allowed Values:

- Yes = PBX will send a confirmation tone.
- No = PBX will not send a confirmation tone.

Default Value = No

INI File Parameter Name = telMwiConfirm

3.9.3.2 Use Same Port for MWI Clear/Set

Description: Specifies if the PBX requires that the telephony station that set an MWI be used to clear the MWI.

Note: Some PBX types require that the port that set an MWI be the same port that clears the MWI.

Allowed Values:

- Yes = Use the same port to set and clear the MWI.
- No = Not necessary to use the same port to clear an MWI that was used to set the MWI.

Default Value = Yes

INI File Parameter Name = telMwiSamePort

3.9.4 CPID Settings Group

The Analog parameters in the CPID Settings group include:

- Initial Wait for Inband CPID
- Inband CPID Complete Timeout
- CID to First Ring Timeout
- Analog Interface Type
- Central Office (Type I) Caller ID Type
- Central Office (Type I) Caller ID Alert Type
- Central Office (Type I) FSK Caller ID Expiration
- Central Office (Type I) FSK Caller ID Timeout
- Auto-Answer Inbound TDM Calls (Type II CPID)

3.9.4.1 Initial Wait for Inband CPID

Description: Specifies the number of milliseconds the DMG1000 will wait for the first inband DTMF digit to arrive after answering an incoming PBX call.

Allowed Values: 100 to 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telInbCpidStartMs

3.9.4.2 Inband CPID Complete Timeout

Description: Specifies the number of milliseconds the DMG1000 will wait for each subsequent inband DTMF digit to arrive. If no digit is received within this time, the CPID is assumed to be complete.

Allowed Values: 100 to 2000 milliseconds

Default Value = 300 milliseconds

INI File Parameter Name = telInbCpidEndMs

3.9.4.3 CID to First Ring Timeout

Description: Specifies the number of milliseconds that are allowed from the end of the Type I CPID on the first ring. If this time is exceeded, the CPID is cleared.

Note: This parameter is only needed when CPID arrives at the Gateway before the first ring. This parameter can be ignored if the CPID arrives between the first and second ring.

Allowed Values: 500 to 30000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telAlgPreRingDtmfTimeoutMs

3.9.4.4 Analog Interface Type

Description: Specifies the type of analog interface to which the DMG1000 is connected.

Allowed Values:

- PBX
- Central Office

Default Value = PBX

INI File Parameter Name = telAlgIfType

3.9.4.5 Central Office (Type I) Caller ID Type

Description: Specifies the Type I Caller ID modulation to use, either Bellcore FSK (North America and Australia), DTMF (parts of Europe, including Sweden, Finland, Denmark, Netherlands, and Iceland), or none. This configuration option is only enabled when the Analog Interface Type is Central Office.

Note: If the Caller ID type is DTMF, then CPID parsing rules will need to be specified. See Chapter 6, "Media Gateway Parsers" for further information on the creation and use of parsing rules.

Allowed Values:

- Bellcore FSK
- DTMF
- None

Default Value = None

INI File Parameter Name = telAlgCoCidType

3.9.4.6 Central Office (Type I) Caller ID Alert Type

Description: Specifies the kind of Type I caller ID alert that is used on the trunk(s) connected to the DMG1000, either pause in ring cycle, ring burst, polarity reversal, or none. This configuration option is only enabled when the Central Office (Type I) Caller ID Type is set to **Bellcore FSK**.

Notes: ***1.*** If the pause in ring cycle alert type is used, the Incoming Rings Before Answer parameter should be set to greater than 1 for the caller ID information to be contained in the outgoing voice over IP (VoIP) message.

2. If an alert is used on the trunk that is not explicitly supported in the list (for example, OSI or a dual tone), use the None setting for this parameter.

Allowed Values:

- Pause in Ring Cycle
- Ring Burst
- Polarity Reversal
- None

Default Value = Pause in Ring Cycle

INI File Parameter Name = telAlgCoCidAlertType

3.9.4.7 Central Office (Type I) FSK Caller ID Expiration

Description: Specifies the duration in milliseconds that FSK Type I Caller ID information stays valid. This configuration option is only enabled when the Central Office Caller ID Type is Bellcore FSK.

Allowed Values: 100 to 60000 milliseconds

Default Value = 10000 milliseconds

INI File Parameter Name = telAlgTypeICidExpMs

3.9.4.8 Central Office (Type I) FSK Caller ID Timeout

Description: Specifies the duration in milliseconds that the digital signal processor (DSP) will perform FSK Type I Caller ID detection after an alert is detected. This configuration option is only enabled when the Central Office Caller ID Type is Bellcore FSK and the Caller ID Alert Type is other than None.

Allowed Values: 500 to 30000 milliseconds

Default Value = 5000 milliseconds

INI File Parameter Name = telAlgFskCidTimeoutMs

3.9.4.9 Auto-Answer Inbound TDM Calls (Type II CPID)

Description: Type II CPID requires that the inbound TDM call be answered by the gateway before the switch will send the inband CPID to the gateway. If this parameter is enabled, then the gateway will answer the call and gather the inband CPID from the switch prior to sending a call-request to the VoIP endpoint. This allows the gateway to send the CPID information as part of the initial call-request to the VoIP endpoint. If this parameter is disabled, then the gateway will not answer the inbound TDM call until after the call-request is sent to the VoIP endpoint and the VoIP endpoint answers the call-request. In this case, the CPID information will not be part of the initial call-request to the VoIP endpoint.

Note: This parameter is valid only when the gateway is configured to receive Type II CPID (post-answer CPID).

Allowed Values:

- Yes = Gateway will auto-answer the inbound TDM call.
- No = Gateway will not auto-answer the inbound TDM call.

Default Value = No

INI File Parameter Name = telAutoAnswer

3.9.5 Rx/Tx Gain Control Group

The Analog parameters in the Rx/Tx Gain Control group include:

- Analog Receive Gain

3.9.5.1 Analog Receive Gain

Description: Specifies an amplification factor (in dB) applied to the received analog signal.

Note: Changing this setting can adversely affect the quality of voice calls. It is recommended that this parameter be changed only after consulting Dialogic support.

Allowed Values:

- 0 dB = no amplification
- 3 dB = amplify by 3 dB
- 6 dB = amplify by 6 dB
- 9 dB = amplify by 9 dB

Default Value = 0 dB

INI File Parameter Name = telAlgAnalogRxGain

3.10 TDM Digital Parameters

The TDM Digital subgroup includes the following parameter:

- Telephony Switch Type

Telephony Switch Type

Description: Specifies the type of PBX telephony switch to which the DMG1000 is connected.

Note: This parameter does not apply to the Models DMG1008MTLDNIW, DMG1008LSW, DMG1004LSW, DMG2030DTIQ, DMG2060DTIQ, and DMG2120DTIQ.

Note: The list of allowed values that will be displayed depends on your model type.

Allowed Values:

- None = No PBX Type has been selected.
- M1 = Nortel Meridian-1/Meridian SL-1
- Norstar = Nortel Norstar Key Systems
- Optiset_300 = Siemens Hicom North American Configuration
- Optiset_300_Europe = Siemens Hicom European Configuration
- Lucent = Lucent DEFINITY G3
- Magix = Merlin Magix
- NEC_IMG = NEC IMG
- NEC_IMX = NEC IMX
- NEC_NEAX = NEC NEAX or NEC 2400 IPX
- Rolm_8000 = Rolm 8000
- Rolm_9751_SW9005 = Rolm 9751 with software release 9005 or earlier
- Rolm_9751_SW9006 = Rolm 9751 with software release 9006 or earlier

Default Value = None

INI File Parameter Name = telPbxType

Note: If Optiset_300_Europe is selected as the Telephony Switch Type, the setting of subscriber Message Waiting Indicator (MWI) lights is not supported and the Analog version of the DMG1000 (Models DMG1008LSW, DMG1004LSW) should be used to set/clear subscriber MWIs on this switch type.

Note: Unit requires a restart if this parameter value is changed.

3.11 TDM Port Enable Parameters

The TDM Port Enable parameters allow you to individually configure the call control capabilities of each port connected to the Media Gateway. These parameters also allow you to enable or disable individual ports. The TDM Port Enable group includes the following parameters:

- Port #
- Telephony Port Enabled

3.11.0.1 Port

Description: This is a read-only parameter that defines the Media Gateway port numbers.

Note: If the T1/E1 signaling mode is configured for ISDN, the Port # refers to the T1 or E1 span number.

3.11.0.2 Telephony Port Enabled

Description: Specifies whether the telephony port is disabled or enabled. Disabled ports drop communications links to the PBX.

Note: If the T1/E1 signaling mode is configured for ISDN, the Port # refers to the T1 or E1 span number.

Allowed Values:

- Yes = Port is enabled and is capable of providing a connection. The LED for this port may or may not be green, depending on whether or not a line is plugged in.
- No = Port is disabled and is incapable of providing a connection. The LED for this port will be red, regardless of whether or not a line is plugged in.

Default Value = Yes

INI File Parameter Name = telPortEnabled

3.12 TDM Call Type Group

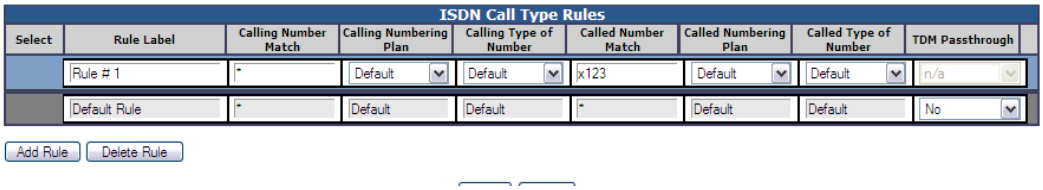
The TDM Call Type group allows the user to specify the type of number for the called and calling party information, which then can be sent to PBX to decide the call type. During a VoIP to ISDN call, this call type information for called and calling party can be indicated to the PBX.

3.12.1 ISDN Call Type Rules

The Web interface contains two independent divisions for called and calling party rules. The management interface will allow user to create multiple rules to set the call type for inbound and outbound call numbers. The user can define rules to assign call type for each calling and called party information for all outbound ISDN calls. Rules are evaluated from the top down until a matching rule is found.

Figure 17 is a screen shot of the ISDN Call Type Rules Web page:

Figure 17. ISDN Call Type Rules Web Page



Select

Use this column to select a rule and change its priority. Click and hold the left mouse button and drag the row up or down.

Rule Label

Specifies name of the call type rule.

Calling Number Match

Each CPID Matching rule must use the following syntax:

Table 2. Syntax for Number Matching

Token	Description
*	Matches all.
0 1 2 3 4 5 6 7 8 9	Identifies a specific digit.
[digit_string-digit_string]	Specifies a range of digit strings.
x	Matches any *single* digit.
.	Matches any number of *ending* digits.

Calling Numbering Plan

Numbering Plan for the calling party information. Initially the value will be default which will set an appropriate value for the number plan depending on the calling number.

Allowed Values:

- Default
- Unknown
- ISDN_E.164
- Data_X.121
- Telex_F.69
- Standard

- Private
- Reserved

Table 3. Default Number Plan

NI-2, DMS-100, 5ESS	
Length = 7	ISDN_E.164
Length = 10	ISDN_E.164
All other lengths	Unknown
QSIG, ETSI	
All number lengths	Unknown

Calling Type of Number

Type of Number for the calling party information. The user must select matching number type according to the selected number plan. If the value is set to default, the gateway will set an appropriate value depending on the calling number.

Allowed Values:

- Default
- Unknown
- International
- National
- Network
- Subscriber
- Abbreviated
- Reserved
- Level2Regional
- Level1Regional
- PISNSpecific
- Level0Regional

Table 4. Default Number Type

NI-2, DMS-100, 5ESS	
Length = 7	Subscriber
Length = 10	International
All other lengths	Unknown
QSIG, ETSI	
All number lengths	Unknown

Called Number Match

Each CPID Matching rule must use the following syntax:

Table 5. Syntax for Number Matching

Token	Description
*	Matches all.
0 1 2 3 4 5 6 7 8 9	Identifies a specific digit.
[digit_string-digit_string]	Specifies a range of digit strings.
x	Matches any *single* digit.
.	Matches any number of *ending* digits.

Called Numbering Plan

Numbering Plan for the called party information. Initially the value will be default which will set an appropriate value for the number plan depending on the called number.

Allowed Values:

- Default
- Unknown
- ISDN_E.164
- Data_X.121
- Telex_F.69
- Standard
- Private
- Reserved

Table 6. Default Number Plan

All	Unknown

Called Type of Number

Type of Number for the called party information. The user must select matching number type according to the selected number plan. If the value is set to default, the gateway will set an appropriate value depending on the called number.

Allowed Values:

- Default
- Unknown
- International
- National
- Network
- Subscriber
- Abbreviated
- Reserved
- Level2Regional
- Level1Regional
- PISNSpecific
- Level0Regional

Table 7. Default Number Type

All	Unknown

TDM Passthrough

Valid only for the Default Rule. If set to Yes and call routing is TDM to TDM, the call type information is copied from inbound to outbound call. Otherwise, the outbound call type information is set to the default values.

Allowed Values:

- Yes
- No

3.13 TDM Cause Codes Group

The TDM Cause Codes group provides tables for mapping ISDN Cause Codes and SIP Responses.

3.13.1 ISDN Cause Code Mapping

The Web interface contains two tables that define mapping between ISDN Cause Codes and SIP Responses.

- ISDN Cause Codes to SIP Response
- SIP Responses to ISDN Cause Codes

3.13.1.1 ISDN Cause Codes to SIP Response

The ISDN to SIP Mapping table maps ISDN Cause Codes to generated SIP Responses. This table takes effect under the following scenarios: on progress messages, disconnect received, disconnect indicator, and when the set facility request fails.

Figure 18 is a screen shot of the ISDN to SIP Mapping Web page:

Figure 18. ISDN to SIP Mapping Web Page

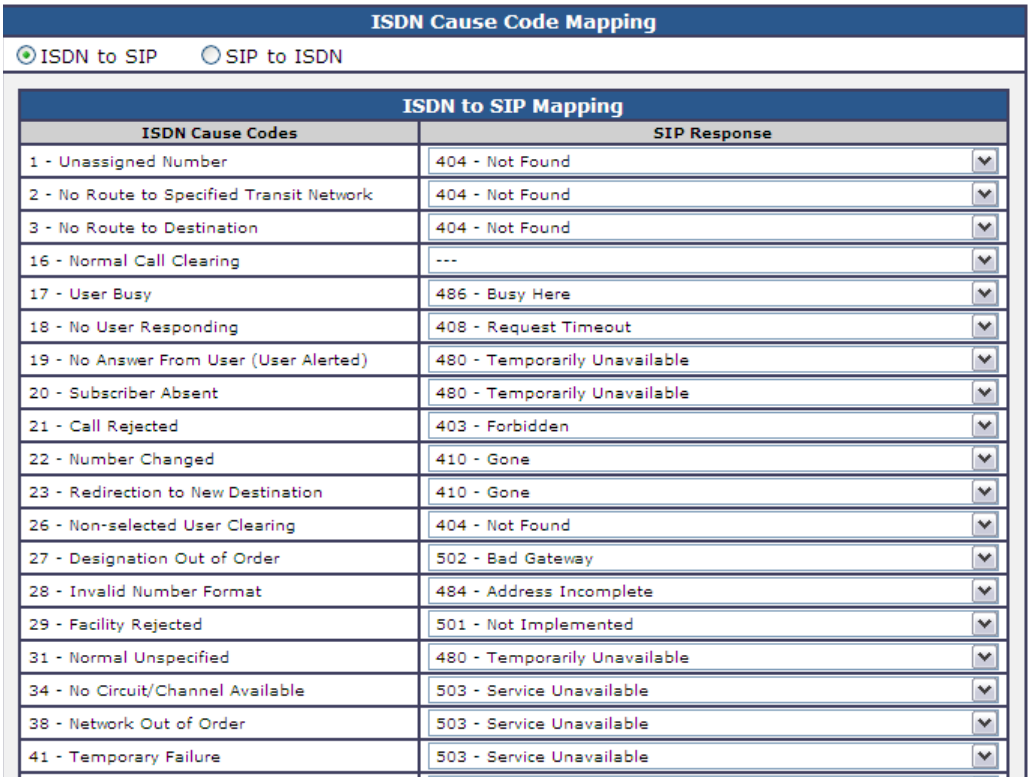

ISDN Cause Code Mapping
ISDN to SIP
SIP to ISDN
ISDN to SIP Mapping
ISDN Cause Codes
SIP Response
1 - Unassigned Number
404 - Not Found
2 - No Route to Specified Transit Network
404 - Not Found
3 - No Route to Destination
404 - Not Found
16 - Normal Call Clearing
---
17 - User Busy
486 - Busy Here
18 - No User Responding
408 - Request Timeout
19 - No Answer From User (User Alerted)
480 - Temporarily Unavailable
20 - Subscriber Absent
480 - Temporarily Unavailable
21 - Call Rejected
403 - Forbidden
22 - Number Changed
410 - Gone
23 - Redirection to New Destination
410 - Gone
26 - Non-selected User Clearing
404 - Not Found
27 - Designation Out of Order
502 - Bad Gateway
28 - Invalid Number Format
484 - Address Incomplete
29 - Facility Rejected
501 - Not Implemented
31 - Normal Unspecified
480 - Temporarily Unavailable
34 - No Circuit/Channel Available
503 - Service Unavailable
38 - Network Out of Order
503 - Service Unavailable
41 - Temporary Failure
503 - Service Unavailable

Table 8 lists the supported SIP Responses.

Table 8. ISDN to SIP Mapping

SIP Responses	
---	Result in the sending of a SIP BYE request
300	Multiple Choices
301	Moved Permanently
302	Moved Temporarily
305	Use Proxy
400	Bad Request
401	Unauthorized
402	Payment Required
403	Forbidden
404	Not Found
405	Method Not Allowed
406	Not Acceptable
407	Proxy Authentication
408	Request Timeout
410	Gone
413	Request Entity Too Long
414	Request URI Too Long
415	Unsupported Media Type
420	Bad Extension
480	Temporarily Unavailable
481	CLEG Does Not Exist
482	Loop Detected
483	Too Many Hops
484	Address Incomplete
485	Ambiguous
486	Busy Here
487	Request Terminated
488	Not Acceptable Here
500	Server Internal Error
501	Not Implemented
502	Bad Gateway
503	Service Unavailable
504	Gateway Timeout

Table 8. ISDN to SIP Mapping

SIP Responses	
505	Version Not Supported
600	Busy Everywhere
603	Decline
604	Does Not Exist Anywhere

3.13.1.2 SIP Responses to ISDN Cause Codes

The SIP to ISDN Mapping table maps SIP Responses to ISDN Cause Codes. This table takes effect under the following scenarios: disconnect request and release request.

Figure 19 is a screen shot of the SIP to ISDN Mapping Web page:

Figure 19. SIP to ISDN Mapping Web Page

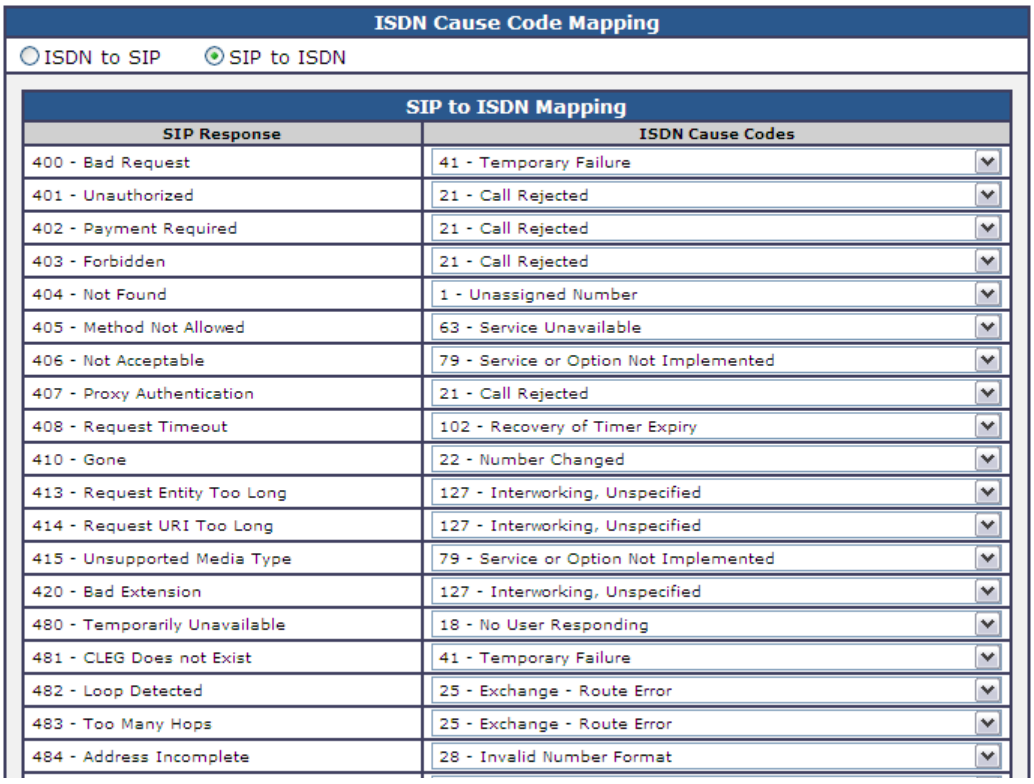

ISDN Cause Code Mapping
ISDN to SIP
SIP to ISDN
SIP to ISDN Mapping
SIP Response
ISDN Cause Codes
400 - Bad Request
41 - Temporary Failure
401 - Unauthorized
21 - Call Rejected
402 - Payment Required
21 - Call Rejected
403 - Forbidden
21 - Call Rejected
404 - Not Found
1 - Unassigned Number
405 - Method Not Allowed
63 - Service Unavailable
406 - Not Acceptable
79 - Service or Option Not Implemented
407 - Proxy Authentication
21 - Call Rejected
408 - Request Timeout
102 - Recovery of Timer Expiry
410 - Gone
22 - Number Changed
413 - Request Entity Too Long
127 - Interworking, Unspecified
414 - Request URI Too Long
127 - Interworking, Unspecified
415 - Unsupported Media Type
79 - Service or Option Not Implemented
420 - Bad Extension
127 - Interworking, Unspecified
480 - Temporarily Unavailable
18 - No User Responding
481 - CLEG Does not Exist
41 - Temporary Failure
482 - Loop Detected
25 - Exchange - Route Error
483 - Too Many Hops
25 - Exchange - Route Error
484 - Address Incomplete
28 - Invalid Number Format

Table 9 lists the supported ISDN Cause Codes.

Table 9. SIP to ISDN Mapping

ISDN Cause Codes		
1	0x1	Unassigned Number
2	0x2	No Route to Specified Transit Network
3	0x3	No Route to Destination
6	0x6	Channel is Unacceptable
7	0x7	Call Awarded and Being Delivered in an Established Channel
16	0x10	Normal Call Clearing
17	0x11	User Busy
18	0x12	No User Responding
19	0x13	No Answer From User - User Alerted
20	0x14	Subscriber Absent
21	0x15	Call Rejected
22	0x16	Number Changed
23	0x17	Redirection to New Destination
25	0x19	Exchange - Routing Error
26	0x1A	Non-Selected User Clearing
27	0x1B	Designation Out of Order
28	0x1C	Invalid Number Format
29	0x1D	Facility Rejected
31	0x1F	Normal Unspecified
34	0x22	No Circuit/Channel Available
38	0x26	Network Out of Order
41	0x29	Temporary Failure
42	0x2A	Switching Equipment Congestion
43	0x2B	Access Information Discarded
44	0x2C	Requested Channel/Circuit Not Available
47	0x2F	Resource Unavailable, Unspecified
55	0x37	Incoming Called Barred Within CUG
57	0x39	Bearer Capability Not Authorized
58	0x3A	Bearer Capability Not Presently Available
63	0x3F	Service Unavailable
65	0x41	Bearer Capability Not Implemented
66	0x42	Channel Type Not Implemented
69	0x45	Requested Facility Not Implemented

Table 9. SIP to ISDN Mapping

ISDN Cause Codes		
70	0x46	Only Restricted Digital Available
79	0x4F	Service or Option Not Implemented
81	0x51	Invalid Call Reference Value
82	0x52	Channel Does Not Exist
83	0x53	A Suspended Call Exists, But This Call Identity Does Not
84	0x54	Call ID In Use
85	0x55	No Call Suspended
86	0x56	Call Having the Requested Call Identity Has Been Cleared
87	0x57	User Not Member of CUG
88	0x58	Incompatible Destination
91	0x5B	Invalid Transit Network Selection
95	0x5F	Invalid Message, Unspecified
96	0x60	Mandatory Information Element is Missing
97	0x61	Message Type Non-existent or Not Implemented
98	0x62	Message Incompatible With Call State or Non-existent
99	0x63	Information Element Not Implemented
100	0x64	Invalid Information Element Contents
101	0x65	Message Not Compatible with Call State
102	0x66	Recovery of Timer Expiry
111	0x6F	Protocol Error, Unspecified
127	0x7F	Interworking, Unspecified

3.14 TDM CPID Parsing Configuration

The TDM CPID Parsing Configuration data is included for T1 CAS in the DMG2000 unit and in the Analog DMG1000 unit. For other models, this data can be accessed instead by navigating to the *TDM-->CPID Parsing* link via the Web interface. Modifying this data affects the display parsing of the Media Gateway.

Caution: The TDM CPID Parsing Configuration data should only be modified by advanced users of the Media Gateway. You are strongly advised to backup your current configuration before proceeding. See Section 2.6.1, "Exporting Configuration Information", on page 41 for additional information about backing up your configuration.

The text box on this page can be easily modified by either pasting parsing rules into it from a text file or directly modifying the contents of the control. See Chapter 6, "Media Gateway Parsers" for further information on the creation and use of parsing rules.

INI File Parameter Name = ;-CPID RULES

3.15 Serial Ports Parameters

The following parameters are in the Serial ports group:

- Serial Port, COM1 Group
- Serial Port, COM2 Group (DMG2000 Only)

3.15.0.1 Serial Port, COM1 Group

The Serial port, COM1 group parameters are used to configure the DMG1000 Serial interface port DIAGNOSTICS connector and the DMG2000 COM1 port connector. The system parameters in the Serial port group include:

- Serial Port Baud Rate
- Serial Port Parity
- Serial Port Data Bits
- Serial Port Stop Bits

Serial Port Baud Rate

Description: Specifies the baud rate of the Media Gateway Serial port.

Allowed Values:

- 1200 = 1200 bps
- 2400 = 2400 bps
- 9600 = 9600 bps
- 19200 = 19200 bps
- 38400 = 38400 bps

Default Value = 38400

INI File Parameter Name = sysSerialBaudRate

Serial Port Parity

Description: Specifies the parity of the Media Gateway Serial port.

Allowed Values:

- None = Parity will not be used.
- Even = Even parity will be used.
- Odd = Odd parity will be used.

Default Value = None

INI File Parameter Name = sysSerialParity

Serial Port Data Bits

Description: Specifies the number of data bits used by the Media Gateway Serial port.

Allowed Values:

- 7 data bits
- 8 data bits

Default Value = 8 data bits

INI File Parameter Name = sysSerialDataBits

Serial Port Stop Bits

Description: Specifies the number of stop bits used by the Media Gateway Serial port.

Allowed Values:

- 1 stop bit
- 2 stop bits

Default Value = 1 stop bit

INI File Parameter Name = sysSerialStopBits

3.15.0.2 Serial Port, COM2 Group (DMG2000 Only)

The Serial port, COM2 group parameters are used to configure the DMG2000 diagnostics/administration Serial port (COM2 connector). The system parameters in the Serial port group include:

- Serial Port Baud Rate
- Serial Port Parity
- Serial Port Data Bits
- Serial Port Stop Bits

Serial Port Baud Rate

Description: Specifies the baud rate of the DMG2000 Serial port.

Allowed Values:

- 1200 = 1200 bps
- 2400 = 2400 bps
- 9600 = 9600 bps
- 19200 = 19200 bps
- 38400 = 38400 bps
- 57600 = 57600 bps
- 115200 = 115200 bps

Default Value = 115200 bps

INI File Parameter Name = sysSerialBaudRateCom2

Serial Port Parity

Description: Specifies the parity of the DMG2000 Serial port.

Allowed Values:

- None = Parity will not be used.
- Even = Even parity will be used.
- Odd = Odd parity will be used.

Default Value = None

INI File Parameter Name = sysSerialParityCom2

Serial Port Data Bits

Description: Specifies the number of data bits used by the DMG2000 Serial port.

Allowed Values:

- 7 data bits
- 8 data bits

Default Value = 8 data bits

INI File Parameter Name = sysSerialDataBitsCom2

Serial Port Stop Bits

Description: Specifies the number of stop bits used by the DMG2000 Serial port.

Allowed Values:

- 1 stop bit
- 2 stop bits

Default Value = 1 stop bit

INI File Parameter Name = sysSerialStopBitsCom2

3.16 Serial Ports Switch Protocol Parameters

Note: The Switch Protocol Web page does not exist for Mitel or Rolm DMG1000 units.

The Serial Protocol parameters are used to define the Serial protocol used in the connection to the PBX Serial link. These parameters include:

- Serial Mode (Master/Slave)
- Serial Interface Protocol
- MCI Message Extension Length
- MCI Message Type
- CPID Length
- CPID Padding String
- Voice Mail Port Length
- System Number
- MWI Response Timeout
- IP Address of Serial Server
- Serial CPID Expiration

3.16.1 Serial Mode (Master/Slave)

Description: Specifies if the Media Gateway is using the Serial protocol and, if so, whether the unit is the Serial Protocol Master or Slave. The Media Gateway must be configured as the Master if it is physically connected to the Serial link of the PBX. In a system with multiple Media Gateways, only one of the units can be connected to the Serial link of the PBX. This unit is the Serial Protocol Master and all other Media Gateways are considered to be the slaves.

Note: The master and slaves use the IP network to communicate Serial protocol information to each other.

Allowed Values:

- None = Serial link is not used.
- Master = This Media Gateway is connected to the PBX Serial link.
- Slave = This Media Gateway is part of a multiple configuration, but is not connected to the PBX Serial link.

Default Value = None

INI File Parameter Name = telSerMode

Note: Unit requires a restart if this parameter value is changed.

3.16.2 Serial Interface Protocol

Description: Sets the Media Gateway to the Serial protocol used by the PBX. Only valid when the Serial Mode parameter is set to Master.

Allowed Values:

- SMDI
- MCI
- MD110

Default Value = SMDI

INI File Parameter Name = telSerProtocol

Note: Unit requires a restart if this parameter value is changed.

3.16.3 MCI Message Extension Length

Description: Specifies the extension length used in MCI messages. Messages with six-digit extensions or with eight-digit extensions.

Allowed Values:

- Six-Digits
- Eight-Digits

Default Value = Six-Digits

INI File Parameter Name = telSerMciMsg

3.16.4 MCI Message Type

Description: Specifies the type of MCI messages. Messages can be Type B (default) or Type A. Type B messages include a tenant number of '01' while Type A messages exclude a tenant number.

Allowed Values:

- Type_A
- Type_B

Default Value = Type_B

INI File Parameter Name = telSerMciMsgType

3.16.5 CPID Length

Description: For SMDI, specifies the length of the extension field in an MWI request sent to the PBX from the DMG1000. For MD110, specifies the length of the calling and called party information contained in the CPID Serial data packet from the PBX.

This parameter is only required when the Serial protocol is set to SMDI or MD110 and the Media Gateway is the Serial Protocol Master.

Allowed Values: 2 to 10

Default Value = 7

INI File Parameter Name = telSerCpidLen

3.16.6 CPID Padding String

Description: Specifies the pad string to strip from the CPID fields (calling and called parties) in the incoming Serial packet data. Also used to pad extensions in MWI requests sent to the PBX from the Media Gateway. This pad string must match the pad string configured by the PBX.

This parameter is only required when the Serial Protocol parameter is set to SMDI and the Media Gateway is the Serial Protocol Master.

Allowed Values: A string of up to 10 digits

Default Value = (no default value)

INI File Parameter Name = telSerCpidPadStr

3.16.7 Voice Mail Port Length

Description: Specifies the length of the field in the Serial data packet that contains the "voice mail port" number that the call arrived (or will arrive) on. This parameter is only required when the Serial protocol is set to MD110 and the Media Gateway is the Serial Protocol Master.

Allowed Values: 2 to 5

Default Value = 2

INI File Parameter Name = telSerVmpLen

3.16.8 System Number

Description: Specifies the "voice mail system" that the Serial protocol packet is being generated from. This parameter is used by the PBX in the event there is more than one application server (e.g. voice mail, unified messaging server, etc.) connected to the PBX. This parameter is only required when the Serial protocol parameter is set to a value of MD110 and this Media Gateway is configured as the Serial Protocol Master.

Allowed Values: 0 to 99

Default Value = 1

INI File Parameter Name = telSerSysNum

3.16.9 MWI Response Timeout

Description: Specifies the time in milliseconds the Media Gateway Serial protocol Master will wait for a failed response from the PBX before sending a success message to a Serial protocol Slave in response to a message waiting indicator (MWI) request from the slave. This parameter is only required when this Media Gateway is configured as the Serial Protocol Master.

Allowed Values: 100 to 60000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telSerMwiRspToutMs

3.16.10 IP Address of Serial Server

Description: Specifies the IP Address of the Media Gateway that is configured as the Serial Protocol Master. Serial Protocol slave devices send and receive all Serial protocol information to/from the Master device via the IP network. This parameter is only required when the Media Gateway is configured as the Serial Protocol Slave.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = telSerAddrSrvr

Note: Unit requires a restart if this parameter value is changed.

3.16.11 Serial CPID Expiration

Description: Specifies the time in milliseconds that Serial CPID information received by a Media Gateway (Master or Slave) remains valid. If the timeout expires before an inbound call is received on the Media Gateway PBX port indicated by the Serial CPID information, the Serial CPID information is discarded.

Allowed Values: 100 to 60000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telSerCpidExpMs

3.17 Tone Detection Parameters

The Tone Detection parameters define the characteristics (frequencies, durations, and deviations) of the tones that the Media Gateway detects during call progress analysis.

These call progress tones can be viewed and edited manually from the Web interface if the characteristics are known, or the Media Gateway can be directed to analyze and learn the characteristics of specific call progress tones. For more information on viewing, editing, learning, and validating call progress tones from the Web interface, see Chapter 4, "Call Progress Tones".

The Tone Detection parameters include the following:

- ID
- Tone Event
- Tone Name
- Cadence Type
- Number of Cadence Cycles
- Tone Frequency
- Tone Frequency Deviation
- Tone Cadence Time
- Tone Cadence Time Deviation

The Tone Generation Configuration parameters include the following:

- Call Progress Tone Generation Event
- Call Progress Tone Generation Name
- Call Progress Tone Generation Num Cadence Cycles
- Call Progress Tone Generation Frequency 1 (Hz)
- Call Progress Tone Generation Frequency 2 (Hz)
- Call Progress Tone Generation Frequency 3 (Hz)
- Call Progress Tone Generation Frequency 4 (Hz)
- Call Progress Tone Generation Amplitude 1 (dBm)
- Call Progress Tone Generation Amplitude 2 (dBm)
- Call Progress Tone Generation Amplitude 3 (dBm)
- Call Progress Tone Generation Amplitude 4 (dBm)
- Call Progress Generation Cadence On Time
- Call Progress Generation Cadence Off Time

In addition, you can edit the INI file directly if needed as discussed in the following topic:

- Editing the INI File Directly

3.17.0.1 ID

Description: Automatically assigned IDs for reference. These IDs cannot be edited.

3.17.0.2 Tone Event

Description: Specifies the tone event for this tone definition.

Allowed Values:

- None
- Dialtone
- Ringback
- Busy
- Congestion
- Disconnect
- Error
- SIT
- Pager
- Modem
- Fax
- FaxCNG

Default Value = None

INI File Parameter Name = cpToneEvent

3.17.0.3 Tone Name

Description: Specifies the text name associated with this tone definition.

Allowed Values: String with length between 0 - 31 characters

Default Value = Empty string

INI File Parameter Name = cpToneName

3.17.0.4 Cadence Type

Description: Specifies the cadence type for this tone definition.

Allowed Values:

- Continuous = an “on” tone with no “off”
- OnOff = an “on” cadence followed by an “off” cadence
- OnOffQuick = detection is reported at the end of the “off” period following the first valid “on” period
- Alternating = two tones that alternate with no “off” time in between
- Sequential = one, two or three tones that come in sequence

Default Value = Continuous

INI File Parameter Name = cpCadenceType

3.17.0.5 Number of Cadence Cycles

Description: Specifies the number of cadence cycles for this tone definition.

Allowed Values: 1-10

Default Value = 2

INI File Parameter Name = cpCadenceNumCycles

3.17.0.6 Tone Frequency

Description: Specifies the frequencies for this tone definition. A tone can be described by up to three (3) frequencies. For Alternating and Sequential Tones, the top frequency is the first frequency, the middle frequency is the second frequency, and the bottom frequency is the third frequency. A value of 0 means to ignore the entry.

Allowed Values: 0-2000 Hz

Default Value = 0

INI File Parameter Name = cpToneFreq

3.17.0.7 Tone Frequency Deviation

Description: Specifies the deviation for the tone frequencies for this tone definition.

Allowed Values: 0-200 Hz

Default Value = 0

INI File Parameter Name = cpToneFreqDeviation

3.17.0.8 Tone Cadence Time

Description: Specifies the cadence times for this tone definition. A tone can be described by up to three (3) cadence times.

The interpretation of each time parameter is dependent on the selection of Cadence Type. For a given cadence type, Cadence Time specifies the time in milliseconds.

- Continuous - The first time value specifies the minimum time on that must elapse before the tone on event is generated.
- OnOff - The first two time values are used to specify the on time and off time respectively.
- OnOffQuick - The first time value is used to specify the minimum on time that causes detection and the second time value specifies the maximum off time that causes a cadence break.
- Alternating - The first two time values specify the on time for the corresponding frequency component.
- Sequential - For each of the tones in the sequence the corresponding time value specifies the on-time for the tone.

Allowed Values: 0-10000 milliseconds

Default Value = 1500 milliseconds

INI File Parameter Name = cpToneTime

3.17.0.9 Tone Cadence Time Deviation

Description: Specifies the deviation for the cadence times for this tone definition.

Allowed Values: 0-1000 milliseconds

Default Value = 700 milliseconds

INI File Parameter Name = cpToneTimeDeviation

3.17.1 Tone Generation Configuration Parameters

3.17.1.1 Call Progress Tone Generation Event

Description: The Tone event or type. The tone will be interpreted as this type of call progress tone generation.

Allowed Values: The configurable tone events are Dialtone, Busy, Ringback, Congestion, Error, Disconnect, Intercept, Reorder, and Special.

INI File Parameter Name = cpGenToneEvent

Note: This is a *.ini file parameter only.

3.17.1.2 Call Progress Tone Generation Name

Description: A text string that describes the generation tone.

Allowed Values: String with length between 0 - 31 characters

INI File Parameter Name = cpGenToneName

Note: This is a *.ini file parameter only.

3.17.1.3 Call Progress Tone Generation Num Cadence Cycles

Description: The number of cycles to play the tone.

Allowed Values: Number between 0 - 255.

INI File Parameter Name = cpGenToneNumCycles

Note: This is a *.ini file parameter only.

3.17.1.4 Call Progress Tone Generation Frequency 1 (Hz)

Description: Frequency 1 contained within the tone.

Allowed Values: Number between 0 – 3000. A value of 0 means to ignore the entry.

INI File Parameter Name = cpGenToneFreqHz1

Note: This is a *.ini file parameter only.

3.17.1.5 Call Progress Tone Generation Frequency 2 (Hz)

Description: Frequency 2 contained within the tone.

Allowed Values: Number between 0 – 3000. A value of 0 means to ignore the entry.

INI File Parameter Name = cpGenToneFreqHz2

Note: This is a *.ini file parameter only.

3.17.1.6 Call Progress Tone Generation Frequency 3 (Hz)

Description: Frequency 3 contained within the tone.

Allowed Values: Number between 0 – 3000. A value of 0 means to ignore the entry.

INI File Parameter Name = cpGenToneFreqHz3

Note: This is a *.ini file parameter only.

3.17.1.7 Call Progress Tone Generation Frequency 4 (Hz)

Description: Frequency 4 contained within the tone.

Allowed Values: Number between 0 – 3000. A value of 0 means to ignore the entry.

INI File Parameter Name = cpGenToneFreqHz4

Note: This is a *.ini file parameter only.

3.17.1.8 Call Progress Tone Generation Amplitude 1 (dBm)

Description: Amplitude for frequency 1.

Allowed Values: Number between -80 - 3.

INI File Parameter Name = cpGenToneAmpDbm1

Note: This is a *.ini file parameter only.

3.17.1.9 Call Progress Tone Generation Amplitude 2 (dBm)

Description: Amplitude for frequency 2.

Allowed Values: Number between -80 - 3.

INI File Parameter Name = cpGenToneAmpDbm2

Note: This is a *.ini file parameter only.

3.17.1.10 Call Progress Tone Generation Amplitude 3 (dBm)

Description: Amplitude for frequency 3.

Allowed Values: Number between -80 - 3.

INI File Parameter Name = cpGenToneAmpDbm3

Note: This is a *.ini file parameter only.

3.17.1.11 Call Progress Tone Generation Amplitude 4 (dBm)

Description: Amplitude for frequency 4.

Allowed Values: Number between -80 - 3.

INI File Parameter Name = cpGenToneAmpDbm4

Note: This is a *.ini file parameter only.

3.17.1.12 Call Progress Generation Cadence On Time

Description: Specifies the on time interval for the cadence element in milliseconds.

Allowed Values: Number between 0 - 10000 milliseconds

INI File Parameter Name = cpGenToneOnTimeMs

Note: This is a *.ini file parameter only.

3.17.1.13 Call Progress Generation Cadence Off Time

Description: Specifies the off time interval for the cadence element in milliseconds.

Allowed Values: Number between 0 - 10000 milliseconds

INI File Parameter Name = cpGenToneOffTimeMs

Note: This is a *.ini file parameter only.

3.17.2 Editing the INI File Directly

The call progress tone parameters can be edited directly in the INI file if desired. In older versions of the INI file, 16 tone definitions were listed regardless of the number of actual tone definitions in use. In these older INI files unused tone definitions would have the cpToneEvent parameter set to none. In newer versions of the INI file only the tone definitions in use are listed.

To delete a tone from the INI file, set the cpToneEvent to none, and that tone will be ignored when the INI file is read.

To add a tone to an INI file, either edit a tone definition whose cpToneEvent is set to none, or add a new tone definition. Then edit or add all the associated tone parameters.

3.18 Certificates Parameters

The Certificates Parameters support Secure Real-time Transport Protocol (SRTP) as well as SIP Transport Layer Security (TLS) and include the following groups:

- Certificate Usage Group
- SRTP Group

3.18.1 Certificate Usage Group

The Security parameters in the Certificate Usage group include:

- TLS Certificate Type
- HTTPS Certificate Type

3.18.1.1 TLS Certificate Type

Description: Determines whether the Media Gateway uses a self-assigned or Certificate Authority signed certificate for SIP Transport Layer Security (TLS).

Allowed Values:

- Self Signed = Use the self-signed TLS certificate
- CA Signed = Use the Certificate Authority-signed TLS certificate

Default Value = Self Signed

INI File Parameter Name = secSipTlsUseSelfSignedCert

Note: Unit requires a restart if this parameter value is changed.

3.18.1.2 HTTPS Certificate Type

Description: Determines whether the Media Gateway uses a self-assigned or CA signed certificate for HTTP over TLS (HTTPS).

Allowed Values:

- Self Signed = Use the self-signed TLS certificate
- CA Signed = Use the CA-signed TLS certificate

Default Value = Self Signed

INI File Parameter Name = secSipTlsUseSelfSignedCert

Note: Unit requires a restart if this parameter value is changed.

3.19 DSP Settings Parameters

The DSP Settings parameters are included in the Configuration menu selections. These parameters can be accessed instead by navigating to the *DSP Settings* link via the Web interface.

Caution: Modifying any of these parameters affects the digital signal processors and significantly changes the operation of the Media Gateway.

Caution: The DSP Settings configuration parameters should only be modified by advanced users of the Media Gateway. You are strongly advised to backup your current configuration before proceeding. See Section 2.6.1, "Exporting Configuration Information", on page 41 for additional information about backing up your configuration.

The DSP Settings group includes the following subgroups:

- DSP Advanced Settings
- T.38 Fax Advanced Settings
- Positive Answer Machine Detection

3.19.1 DSP Advanced Settings

The DSP Advanced Settings parameters include:

- TDM to IP Gain Adjustment (DMG1000 Only)
- IP to TDM Gain Adjustment (DMG1000 Only)
- Line Echo Cancellation (DMG1000 Only)
- Line Echo Cancellation NLP (DMG1000 Only)
- Voice Activity Noise Floor
- Voice Activity Measurement Period (DMG2000 Only)
- Voice Activity Signal to Noise Ratio (DMG2000 Only)
- Call Progress Filter Threshold
- Call Progress Filter Debounce
- Call Progress Filter Percent (DMG1000 Only)
- Call Progress Filter Low Cutoff (DMG1000 Only)
- Call Progress Filter High Cutoff (DMG1000 Only)
- Call Progress Filter SNR in dB (DMG2000 Only)
- Call Progress Filter Two Tones Max Twist in dB (DMG2000 Only)
- Jitter-Buffer Minimum Delay (DMG2000 Only)
- Jitter-Buffer Maximum Delay (DMG2000 Only)
- Jitter-Buffer Initial Delay (DMG2000 Only)
- Jitter-Buffer Adaptation Period (DMG2000 Only)
- Jitter-Buffer Deletion Threshold (DMG2000 Only)
- Jitter-Buffer Frame Deletion Mode (DMG2000 Only)
- TDM to TDM Media Always Clear Mode (DMG2000 Only)
- VoIP Packet Loss Concealment (DMG2000 Only)
- IP to TDM AGC Enable (DMG2000 Only)
- IP to TDM AGC Slew Rate (DMG2000 Only)
- IP to TDM AGC Target Level (DMG2000 Only)
- IP to TDM AGC Max Gain (DMG2000 Only)
- IP to TDM AGC Min Gain (DMG2000 Only)
- TDM to IP AGC Enable (DMG2000 Only)
- TDM to IP AGC Slew Rate (DMG2000 Only)
- TDM to IP AGC Target Level (DMG2000 Only)
- TDM to IP AGC Max Gain (DMG2000 Only)
- TDM to IP AGC Min Gain (DMG2000 Only)

3.19.1.1 TDM to IP Gain Adjustment (DMG1000 Only)

Description: Adjusts the gain of the audio signal in the Telephony-to-IP direction.

Allowed Values: -14 dB to +14 dB

Default Value = 0 dB

INI File Parameter Name = dspPbxToIPGain

3.19.1.2 IP to TDM Gain Adjustment (DMG1000 Only)

Description: Adjusts the gain of the audio signal in the IP-to-Telephony direction.

Allowed Values: -14 dB to +14 dB

Default Value = 0 dB

INI File Parameter Name = dspIPtoPbxGain

3.19.1.3 Line Echo Cancellation (DMG1000 Only)

Description: Enables or disables the echo canceller on the PBX (TDM) side.

Allowed Values:

- On = Enables the echo canceller.
- Off = Disables the echo canceller.

Default Value = On

INI File Parameter Name = dspEc

3.19.1.4 Line Echo Cancellation NLP (DMG1000 Only)

Description: Enables or disables the echo canceller non-linear processor (NLP) on the PBX (TDM) side.

Allowed Values:

- On = Enables the echo canceller NLP.
- Off = Disables the echo canceller NLP.

Default Value = On

INI File Parameter Name = dspEcNlp

3.19.1.5 Voice Activity Noise Floor

Description: Defines the noise floor for the voice activity detector (VAD). Signal levels below the value selected will be treated as silence by the VAD.

Allowed Values: -80 dB to -10 dB

Default Value = -40 dB (DMG1000)
-32 dB (DMG2000)

INI File Parameter Name = vadNoiseFloor

Note: Unit requires a restart if this parameter value is changed.

3.19.1.6 Voice Activity Measurement Period (DMG2000 Only)

Description: Specifies the Measurement Period for the Voice Activity Detector. The Voice Activity Detector will wait at least this long to determine whether the signal on the line is voice.

Allowed Values: 20 to 200

Default Value = 30

INI File Parameter Name = vadMeasurementPeriod

3.19.1.7 Voice Activity Signal to Noise Ratio (DMG2000 Only)

Description: Specifies the Signal to Noise Ration (SNR) for the Voice Activity Detector. Tonal signals with an SNR greater than this value will be treated as tones and ignored by the Voice Activity Detector.

Allowed Values: 10 to 20

Default Value = 18

INI File Parameter Name = vadMinSNR

3.19.1.8 Call Progress Filter Threshold

Description: Defines the lower threshold for the Call Progress Tone Detector. Call progress signals below this value will be ignored.

Allowed Values: -38 dB to -20 dB

Default Value = -30 dB

INI File Parameter Name = cpFilterThreshold

Note: Unit requires a restart if this parameter value is changed.

3.19.1.9 Call Progress Filter Debounce

Description: Specifies the debounce time in milliseconds for the Call Progress Tone Detector. This value defines the minimum time that a call progress signal must be present before the detector is triggered.

Allowed Values: 100 to 32768 milliseconds

Default Value = 100 milliseconds

INI File Parameter Name = cpFilterDebounce

Note: Unit requires a restart if this parameter value is changed.

3.19.1.10 Call Progress Filter Percent (DMG1000 Only)

Description: Specifies the percent ratio between tone power and total power for the Call Progress Tone Detector. Call progress tone signals whose power is a smaller percentage of total power than this value will be ignored.

Allowed Values: 25 to 87%

Default Value = 45%

INI File Parameter Name = cpFilterPercent

Note: Unit requires a restart if this parameter value is changed.

3.19.1.11 Call Progress Filter Low Cutoff (DMG1000 Only)

Description: Specifies the low cut-off frequency for the Call Progress Tone Detector. Call progress signals below this frequency will be ignored.

Allowed Values: 250 Hz to 1000 Hz

Default Value = 300 Hz

INI File Parameter Name = cpFilterLowCuttoff

Note: Unit requires a restart if this parameter value is changed.

3.19.1.12 Call Progress Filter High Cutoff (DMG1000 Only)

Description: Specifies the high cut-off frequency for the Call Progress Tone Detector. Call progress signals above this frequency will be ignored.

Allowed Values: 500 Hz to 2000 Hz

Default Value = 650 Hz

INI File Parameter Name = cpFilterHighCutoff

Note: Unit requires a restart if this parameter value is changed.

3.19.1.13 Call Progress Filter SNR in dB (DMG2000 Only)

Description: Specifies the Signal to Noise Ratio (SNR) of the Call Progress Tone Detector. Call progress signals whose SNR is lower than specified will be ignored.

Allowed Values: 0 dB to 40 dB

Default Value = 20 dB

INI File Parameter Name = cpFilterSnr

Note: Unit requires a restart if this parameter value is changed.

3.19.1.14 Call Progress Filter Two Tones Max Twist in dB (DMG2000 Only)

Description: Specifies the maximum twist for two tone Call Progress Tones. Call progress signals that consist of two tones whose power ratio of the tones is greater than the specified twist will be ignored.

Allowed Values: 0 dB to 12 dB

Default Value = 3 dB

INI File Parameter Name = cpFilterMaxTwist

Note: Unit requires a restart if this parameter value is changed.

3.19.1.15 Jitter-Buffer Minimum Delay (DMG2000 Only)

Description: Specifies the minimum jitter-buffer delay. The smaller the delay, the smaller the jitter-buffer. It is recommended that this value be at least the duration of one RTP packet interval. If the minimum, maximum, and initial jitter-buffer delays are set to the same value, the jitter-buffer will operate in the non-adaptive mode.

Allowed Values: 0 to 200 milliseconds

Default Value = 20 milliseconds

INI File Parameter Name = dspJbDelayMinMs

3.19.1.16 Jitter-Buffer Maximum Delay (DMG2000 Only)

Description: Specifies the maximum jitter-buffer delay. A larger delay provides protection against larger network jitter, but increases audio delay. It is recommended that this value be set to at least four times the duration of one RTP packet interval. If the minimum, maximum, and initial jitter-buffer delays are set to the same value, the jitter-buffer will operate in the non-adaptive mode.

Allowed Values: 0 to 200 milliseconds

Default Value = 200 milliseconds

INI File Parameter Name = dspJbDelayMaxMs

3.19.1.17 Jitter-Buffer Initial Delay (DMG2000 Only)

Description: Specifies the starting jitter-buffer delay. It is recommended that this value be set to at least two times the duration of one RTP packet interval. If the minimum, maximum, and initial jitter-buffer delays are set to the same value, the jitter-buffer will operate in the non-adaptive mode.

Allowed Values: 0 to 200 milliseconds

Default Value = 20 milliseconds

INI File Parameter Name = dspJbDelayInitMs

3.19.1.18 Jitter-Buffer Adaptation Period (DMG2000 Only)

Description: Controls the speed at which the jitter-buffer adapts downward when network conditions allow. The larger the value, the slower the jitter-buffer adapts downward when jitter decreases.

Allowed Values: 1000 to 65535 milliseconds

Default Value = 10000 milliseconds

INI File Parameter Name = dspJbAdaptationPeriodMs

3.19.1.19 Jitter-Buffer Deletion Threshold (DMG2000 Only)

Description: When the jitter-buffer grows past this value, frames exceeding the threshold are deleted immediately.

Allowed Values: 0 to 500 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = dspJbDeletionThresholdMs

3.19.1.20 Jitter-Buffer Frame Deletion Mode (DMG2000 Only)

Description: Determines how frames are deleted when the jitter-buffer adapts downward.

Allowed Values:

- Soft = More emphasis on audio quality, but maximum delay may be exceeded.
- Hard = Hard cap on the maximum delay, which may negatively impact audio quality.

Default Value = Soft

INI File Parameter Name = dspJbDeletionMode

3.19.1.21 TDM to TDM Media Always Clear Mode (DMG2000 Only)

Description: Enables media passing between TDM to TDM connections to be transmitted in clear mode. This prevents the Media Gateway from encoding the media which allows analog modems, digital modems and other analog devices to transmit data with bit integrity. However, this means that no in-band call progress detection and generation are available. Most T1 CAS deployments require in-band call progress detection so this setting should be Off for T1 CAS. Some ISDN deployments may rely on in-band call progress detection and generation so this setting may need to be Off in these cases.

Allowed Values:

- On
- Off

Default Value = Off

INI File Parameter Name = dspTdmToTdmAllClearMode

3.19.1.22 VoIP Packet Loss Concealment (DMG2000 Only)

Description: Enables packet loss concealment (PLC). The gateway tries to account for missed media packets. It may attempt generating voice samples based on prior packets for short audio drops. When used in conjunction with IVR systems, PLC may cause unwanted artifacts at the end of voice prompts, in which case PLC should be turned Off.

Allowed Values:

- On
- Off

Default Value = On

INI File Parameter Name = dspPktLossConclmt

3.19.1.23 IP to TDM AGC Enable (DMG2000 Only)

Description: Enables automatic gain control (AGC) in the IP to TDM direction.

Allowed Values:

- On = AGC is enabled.
- Off = AGC is disabled.

Default Value = On

INI File Parameter Name = dspIpToPbxAgcEnable

3.19.1.24 IP to TDM AGC Slew Rate (DMG2000 Only)

Description: Determines the speed at which the AGC adapts in the IP to TDM direction.

Allowed Values:

- Fast = More emphasis on AGC gain adjustments
- Medium = Mid point
- Slow = More emphasis on Audio Quality

Default Value = Medium

INI File Parameter Name = dspIpToPbxAgcSlewRate

3.19.1.25 IP to TDM AGC Target Level (DMG2000 Only)

Description: Defines the IP to TDM AGC target level for a range of -50 to 0 dBm.

Allowed Values: -50 to 0 dBm

Default Value = -14 dBm

INI File Parameter Name = dspIpToPbxAgcLevel

3.19.1.26 IP to TDM AGC Max Gain (DMG2000 Only)

Description: Defines the IP to TDM AGC maximum gain within the range of 0 to 15 dB.

Allowed Values: 0 to 15 dB

Default Value = 12 dB

INI File Parameter Name = dspIpToPbxAgcMaxGain

3.19.1.27 IP to TDM AGC Min Gain (DMG2000 Only)

Description: Defines the IP to TDM AGC minimum gain within the range of 0 to -15 dB.

Allowed Values: 0 to -15 dB

Default Value = -15 dB

INI File Parameter Name = dspIpToPbxAgcMinGain

3.19.1.28 TDM to IP AGC Enable (DMG2000 Only)

Description: Enables automatic gain control (AGC) in the TDM to IP direction.

Allowed Values:

- On = AGC is enabled.
- Off = AGC is disabled.

Default Value = Off

INI File Parameter Name = dspPbxToIpAgcEnable

3.19.1.29 TDM to IP AGC Slew Rate (DMG2000 Only)

Description: Determines the speed at which the AGC adapts in the TDM to IP direction.

Allowed Values:

- Fast = More emphasis on AGC gain adjustments
- Medium = Mid point
- Slow = More emphasis on Audio Quality

Default Value = Medium

INI File Parameter Name = dspPbxToIpAgcSlewRate

3.19.1.30 TDM to IP AGC Target Level (DMG2000 Only)

Description: Defines the TDM to IP AGC target level for a range of -50 to 0 dBm.

Allowed Values: -50 to 0 dBm

Default Value = -14 dBm

INI File Parameter Name = dspPbxToIpAgcLevel

3.19.1.31 TDM to IP AGC Max Gain (DMG2000 Only)

Description: Defines the TDM to IP AGC maximum gain within the range of 0 to 15 dB.

Allowed Values: 0 to 15 dB

Default Value = 12 dB

INI File Parameter Name = dspPbxToIpAgcMaxGain

3.19.1.32 TDM to IP AGC Min Gain (DMG2000 Only)

Description: Defines the TDM to IP AGC minimum gain within the range of 0 to -15 dB.

Allowed Values: 0 to -15 dB

Default Value = -15 dB

INI File Parameter Name = dspPbxToIpAgcMinGain

3.19.2 T.38 Fax Advanced Settings

The T.38 Fax Advanced Settings parameters include:

- Allow T.38 ECM Faxes (DMG2000 Only)
- Transmit Small T4 ECM T.38 Packets (DMG2000 Only)
- Enable T.38 Spoofing (DMG2000 Only)
- TSI Removal (DMG2000 Only)
- CSI Removal (DMG2000 Only)
- NSF Removal (DMG2000 Only)
- DIS Removal (DMG2000 Only)
- T.38 Packet Loss Concealment Method (DMG2000 Only)
- T.38 UDPTL Redundancy Count for Fax Page Data (DMG2000 Only)
- T.38 UDPTL Redundancy Count for T.30 Messages (DMG2000 Only)
- Fax Transmit Level (dBm) (DMG2000 Only)
- Fax TDM Drop Out Bridge Time (DMG2000 Only)
- Fax Modem Carrier Detect Threshold (DMG1000 Only)

3.19.2.1 Allow T.38 ECM Faxes (DMG2000 Only)

Description: Enables/disables Error Correction Mode faxes in T.38.

Allowed Values:

- On = ECM allowed
- Off = ECM disabled

Default Value = On

INI File Parameter Name = dspT38EcmEnable

3.19.2.2 Transmit Small T4 ECM T.38 Packets (DMG2000 Only)

Description: Enables/disables small T4 ECM packet transmission in T.38.

Allowed Values:

- On = Enable small (<256 byte) T4 ECM packet transmission.
- Off = Wait for complete HDLC ECM frame from PCM before processing.

Default Value = On

INI File Parameter Name = dspT38EcmSmallPacketEnable

3.19.2.3 Enable T.38 Spoofing (DMG2000 Only)

Description: Enables/disables spoofing in T.38.

Allowed Values:

- On = Enable T.38 spoofing.
- Off = Disable T.38 spoofing.

Default Value = On

INI File Parameter Name = dspT38SpoofingEnable

3.19.2.4 TSI Removal (DMG2000 Only)

Description: Determines if the Transmitting Subscriber Identification (TSI) is removed.

Allowed Values:

- On - Remove TSI from TDM to reduce packet delay.
- Off - Include TSI in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38TsiRemovalEnable

3.19.2.5 CSI Removal (DMG2000 Only)

Description: Determines if the Called Subscriber Identification (CSI) is removed.

Allowed Values:

- On - Remove CSI from TDM to reduce packet delay.
- Off - Include CSI in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38CsiRemovalEnable

3.19.2.6 NSF Removal (DMG2000 Only)

Description: Determines if Non-Standard Facilities (NSF) is removed.

Allowed Values:

- On - Remove NSF from TDM to reduce packet delay.
- Off - Include NSF in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38NsfRemovalEnable

3.19.2.7 DIS Removal (DMG2000 Only)

Description: Determines if the Digital Identification signal (DIS) is removed.

Allowed Values:

- On - Remove DIS from TDM to reduce packet delay.
- Off - Include DIS in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38DisRemovalEnable

3.19.2.8 T.38 Packet Loss Concealment Method (DMG2000 Only)

Description: Selects the packet loss concealment method in T.38.

Allowed Values:

- None - No T.38 Packet Concealment.
- White_Line - Replace bad line with white line.
- Last_Good_Line - Replace bad line with last good line.

Default Value = White_Line

INI File Parameter Name = dspT38PacketLossConcealment

3.19.2.9 T.38 UDPTL Redundancy Count for Fax Page Data (DMG2000 Only)

Description: Specifies the number of redundant IFP frames transferred with each UDPTL packet.

Allowed Values: 0 to 3

Default Value = 2

INI File Parameter Name = dspUdptlDataRedundancy

3.19.2.10 T.38 UDPTL Redundancy Count for T.30 Messages (DMG2000 Only)

Description: Specifies the number of redundant IFP frames transferred with each UDPTL packet.

Allowed Values: 0 to 3

Default Value = 2

INI File Parameter Name = dspUdptlT30Redundancy

3.19.2.11 Fax Transmit Level (dBm) (DMG2000 Only)

Description: Set the fax transmit level in T.38 fax relay mode.

Allowed Values: -15 to 0

Default Value = -10

INI File Parameter Name = dspT38FaxTransmitLevel

3.19.2.12 Fax TDM Drop Out Bridge Time (DMG2000 Only)

Description: Sets the maximum TDM drop out time in milliseconds that the fax modem will try to bridge.

Allowed Values: 0 to 1200 ms

Default Value = 0 (Disabled)

INI File Parameter Name = dspFaxTdmDropBridgeTime

Note: This parameter is only supported for V.34 release.

3.19.2.13 Fax Modem Carrier Detect Threshold (DMG1000 Only)

Description: Specifies the Threshold used to detect fax carrier.

Allowed Values:

- -26dBm
- -33dBm
- -43dBm

Default Value = -43dBm

INI File Parameter Name = dspT38CDThresholdDbm

Note: This is a *.ini file parameter only.

3.19.3 Positive Answer Machine Detection

The Positive Answer Machine Detection parameters include:

- Maximum Live Answer Time
- Minimum Live Answer Time
- Maximum Silence after Voice has been Detected
- Maximum Time to Wait for Voice

3.19.3.1 Maximum Live Answer Time

Description: Sets the time limit at which point a continuous voice answer will be considered an answer machine.

Allowed Values: 1000 to 20000 milliseconds

Default Value = 2600 milliseconds

INI File Parameter Name = pamdMaxAnswerSize

3.19.3.2 Minimum Live Answer Time

Description: Specifies voice answers have to be at least this long in length to be consider voiced and not noise.

Allowed Values: 10 to 1000 milliseconds

Default Value = 120 milliseconds

INI File Parameter Name = pamdMinAnswerSize

3.19.3.3 Maximum Silence after Voice has been Detected

Description: Specifies any silence that last this long after voice has been detected will be considered an end of voice. If this length of silence is encountered before the maximum live answer time expires, the voice answer will be considered a live person.

Allowed Values: 10 to 10000 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = pamdMaxSilence

3.19.3.4 Maximum Time to Wait for Voice

Description: Sets the maximum amount of time to wait for voice detection before declaring call answered by unknown.

Allowed Values: 1000 to 20000 milliseconds

Default Value = 4000 milliseconds

INI File Parameter Name = pamdFailSafeTimeout

3.20 Non-Menu (Hidden) Parameters

The following parameters do not appear in the Web interface and can only be changed by editing the configuration file directly. These parameters are listed alphabetically.

- DNS NAPTR and SRV Record Enable
- Incompatible Message STATUS
- Inform On No PBX CPID (Phone Emulating Only)
- Inform On No PBX CPID Time (Phone Emulating Only)
- ISDN Overlap Receive Minimum Digits
- ISDN Overlap Receive Timeout
- Routing Proxy Group Name
- Routing Table Only On Proxy Down
- Port Disable Timeout

3.20.1 DNS NAPTR and SRV Record Enable

Description: The gateway can use three types of DNS lookup requests: NAPTR, SRV, and A Record. Typically, the NAPTR and SRV lookup requests do not result in any useful data. To reduce network traffic, the gateway defaults to A record only lookup requests. This parameter can be used to force the gateway to request NAPTR and SRV lookup requests.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = the gateway requests DNS lookup of NAPTR and SRV records.
- No = the gateway does not request DNS lookup of NAPTR and SRV records.

Default Value = No

INI File Parameter Name = sipDnsNaptrSrvLookupEnabled

3.20.2 Incompatible Message STATUS

Description: If 'Yes' then a STATUS message (with cause #98) will be sent if an incompatible message (for the current call state) is received. If 'No' then a STATUS message will not be sent. Applicable only when ISDN Protocol is QSIG or ETSI.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes
- No

Default Value = No

INI File Parameter Name = isdnIncompatibleMsgStatus

3.20.3 Inform On No PBX CPID (Phone Emulating Only)

Description: When this parameter is enabled ('Yes'), the gateway will generate an informational VoIP message to the IP peer if no CPID information is available for the call within the prescribed time-out.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = Media Gateway will generate this informational message.
- No = Media Gateway will not generate this informational message.

Default Value = No

INI File Parameter Name = gwInformIfNoPbxCpid

3.20.4 Inform On No PBX CPID Time (Phone Emulating Only)

Description: Specifies the time in milliseconds that the gateway will send the informational VoIP message indicating that no CPID is available for the call.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 1000 to 60000 milliseconds

Default Value = 4000 milliseconds

INI File Parameter Name = gwInformIfNoPbxCpidMs

3.20.5 ISDN Overlap Receive Minimum Digits

Description: Specifies the minimum number of called-party digits that must be received on an inbound call before the inbound call is processed. If the value is 0, then the call is processed immediately. Applicable only when the switch is using overlap sending.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 0 to 64 digits

Default Value = 0

INI File Parameter Name = isdnOverlapRcvMinDigits

3.20.6 ISDN Overlap Receive Timeout

Description: Specifies the number of milliseconds before the call is answered when in overlap receive mode.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 0 to 65535 milliseconds

Default Value = 14000 milliseconds

INI File Parameter Name = isdnOverlapRcvToutMs

3.20.7 ISDN Service Class

Description: This parameter can be used to control the Bearer Channel Parameters.

Allowed Values:

- Speech
- 3.1_kHz Audio
- Telephony
- FAX_Group_2_and_3

Default Value = Speech

INI File Parameter Name = isdnServiceClass

3.20.8 Routing Proxy Group Name

Description: Specifies the routing proxy group name.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 0 to 39 characters

Default Value = blank

INI File Parameter Name = gwRoutingProxyGroupName

3.20.9 Routing Table Only On Proxy Down

Description: If a VoIP proxy server is available, then all inbound TDM calls and messages will be routed to the VoIP proxy server. If all VoIP proxy servers are unreachable, then the outbound route will be determined by the routing table. All inbound TDM calls and messages will use the routing table to determine the outbound route.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes
- No

Default Value = No

INI File Parameter Name = gwRoutingTableOnlyOnProxyDown

3.20.10 Port Disable Timeout

Description: Represents the time that every port will wait before getting disabled upon restart.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 0 to 10000 ms

Default Value = 0

INI File Parameter Name = telPortdisableTimeout

Note: Changing this value from the default (0) will cause a delay in enabling or disabling of ports and restarting of the gateway.

Call Progress Tones 4

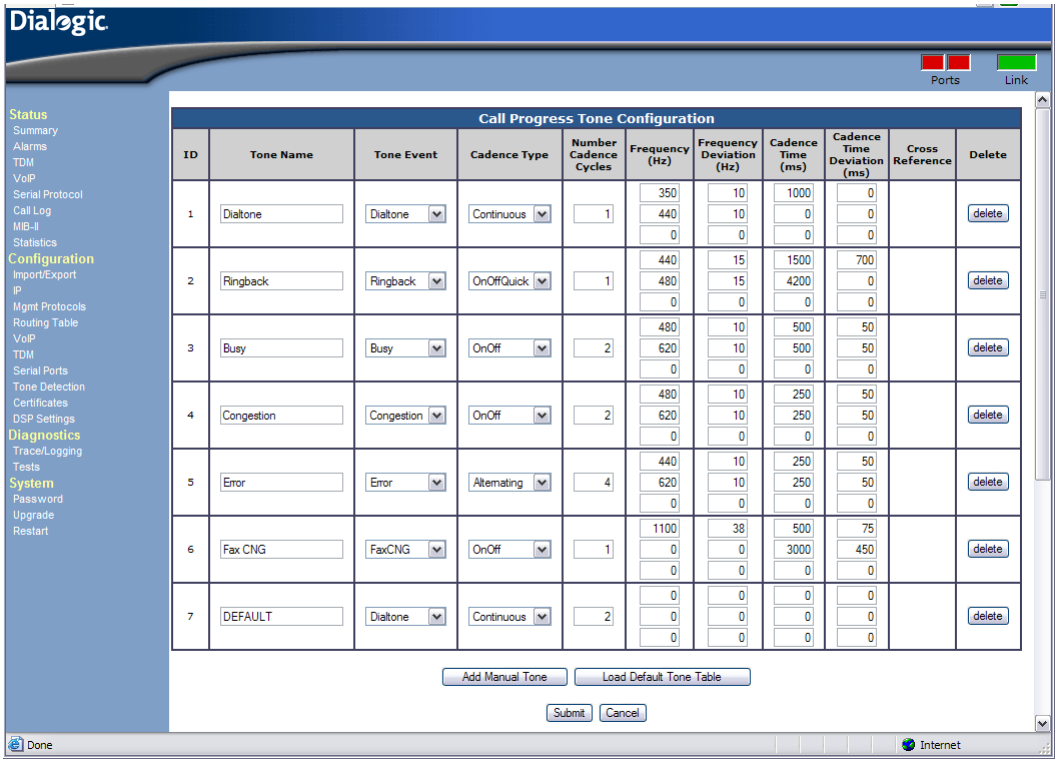
Call progress tone parameters define the characteristics (frequencies, durations, and deviations) of the tones that the Dialogic® Media Gateway detects during call progress analysis. The following sections discuss how to view, edit, learn, and validate call progress tones using a Web interface:

- Viewing and Editing Call Progress Tones 183
- Learning and Validating Call Progress Tones 184

4.1 Viewing and Editing Call Progress Tones

If call progress tone characteristics are known, call progress tones can be viewed and edited manually from the Manual Tones Web page shown in Figure 20. Or the Media Gateway can be directed to analyze and learn the characteristics of specific call progress tones as described in Section 4.2, "Learning and Validating Call Progress Tones", on page 184.

Figure 20. Manual Tones Web Page



You can also add and delete call progress tones from this Web page.

Use the following buttons on the Manual Tones Web page to manipulate call progress tones:

Apply Changes button
Saves changes to the configuration once the parameters have been edited.

Add button
Adds a row to the tone table, allowing a new tone definition to be added.

Reset to Default Values button
Deletes all existing tones and restores factory default tones.

Export Tones Definitions button
Exports the current tone definitions to an INI file. This INI file can then be imported into other Media Gateway, which will update only the call progress tones of the gateway. The tones will also be exported as part of the system configuration information.

For details on the call progress tone parameters, see Section 3.17, "Tone Detection Parameters", on page 154.

4.2 Learning and Validating Call Progress Tones

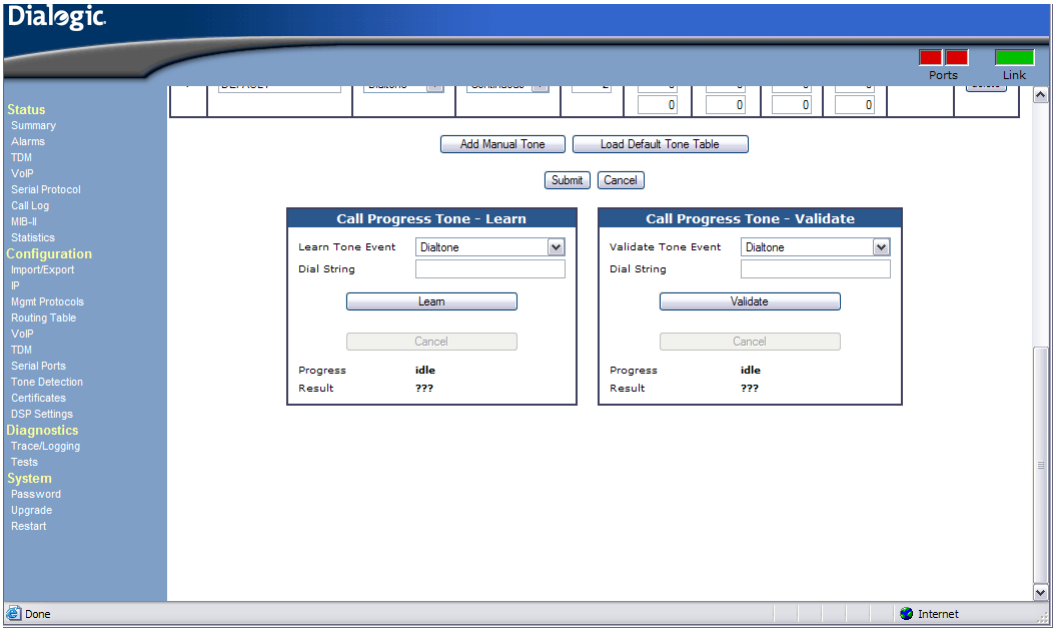
Learning and validating a call progress tone is discussed in the following topics:

- Learn Tone Web Page
- Learning the Characteristics of Unknown Call Progress Tones
- Learn Tone Progress
- Learn Tone Results
- Validating Call Progress Tones
- Validate Tone Progress
- Validate Tone Results

4.2.1 Learn Tone Web Page

The Learn Tone Web page shown in Figure 21 allows you to acquire the characteristics for new call progress tones from the telephony system or to validate the defined call progress tones.

Figure 21. Learn Tone Web Page



The Learn Tone user interface has the following components of interest:

Acquire Tone field
: A check indicates that the corresponding tone event is to be learned or validated.

Tone Event field
: Indicates one of four types: dialtone, busy, error, or ringback.

Destination Address field
: Specifies the phone number or extension that when dialed will generate the call progress tone associated with the tone event. If left blank, the tone will be learned or validated by going off hook. Destination addresses cannot be duplicated in the same session.

Note: It is recommended that you dial each of the destination addresses from a separate phone to validate that the expected call progress tones to be learned or validated are produced.

The following shows possible ways in which call progress tones can be produced:

- Dialtone – For a system dialtone, leaving the destination address blank will allow the unit to determine the dialtone by simply going off hook. For a secondary dialtone, enter the access code for the secondary dialtone in the destination address.
- Busy – Call an extension or number of a phone that is in use. **Caution:** Make sure that the extension is not forwarded on busy and that a busy tone is generated when the extension is dialed.
- Error – Call an invalid extension.

- Ringback – Call a valid extension not in use. **Caution:** Make sure that this extension is not forwarded on "no answer" and that it will ring until answered.

Learn button
: Specifies that the Media Gateway will automatically learn the tone. Used when the frequency and timing information of a call progress tone are unknown. For more information on using the Learn feature, see Section 4.2.2, "Learning the Characteristics of Unknown Call Progress Tones", on page 186.

Validate Only button
: Specifies that the Media Gateway will validate a call progress tone whose frequency and timing information are known. For more information on using the Validate only feature, see Section 4.2.5, "Validating Call Progress Tones", on page 189.

4.2.2 Learning the Characteristics of Unknown Call Progress Tones

If the frequency and timing information of a call progress tone is unknown, use the Learn button to have the Media Gateway automatically learn the tone. This is accomplished by selecting the tone, entering a destination address where the desired call progress tone can be heard and selecting Learn. The unit will then call each destination address and analyze the tones on the line. The result will be a tone definition for the specified tone event that can be added to the tone table.

Follow these steps to learn the timing and frequency characteristics of unknown call progress tones.

1. For each call progress tone that is to be learned, determine how the tone can be generated by the system.

2. Verify that the system will generate the desired call progress tone by dialing that destination address from an extension and listening to verify the tone.

3. Navigate to the Learn Tone Web page (see Figure 21), select the desired tones, and enter the destination addresses for the tones to be learned.

4. Click on the Learn button.

5. Wait for the Learn Tone Results Web page to be displayed. A progress page is displayed showing progress messages as the system proceeds.

6. Select the newly learned tones to be added to the tone definition table and click on the Apply button.

4.2.3 Learn Tone Progress

Once the Learn button has been selected, the Media Gateway will start the process of acquiring the call progress tone characteristics. A Learn Tone Progress page will automatically appear while the Gateway is acquiring the tones and progress messages will be posted.

Progress messages are described as follows:

Learning
The Gateway has gone off hook and dialed the destination address. It is collecting raw data on the call progress tone.

Analyzing
The Gateway is analyzing the raw data to extract the tone characteristics.

Comparing
The Gateway is comparing the new tones to existing tones and other new tones for possible conflicts.

Validating
The Gateway is redialing the destination addresses and testing to see if the new call progress tone definitions can detect the tones.

The Cancel button cancels the current session and returns to the Learn Tone Web page.

4.2.4 Learn Tone Results

When the Media Gateway has finished learning the tones, a results page is displayed.

The learned tones are grouped into the following possible tone results on the Web page:

- Tone Errors
- Existing Tones
- Unique Tones
- Conflicting Tones

After selecting the tones to be added to the tone definition table, the Apply button adds the selected tones to the tone definition table. The Manual Tones Web page discussed in Section 4.1, "Viewing and Editing Call Progress Tones", on page 183 can then be used to view the newly learned tones.

Tone Errors

An error occurred when trying to learn the tones listed in this section. These tones cannot be added to the configuration due to the error encountered. Possible reasons for error as well as possible action include the following:

No data
The Gateway dialed the destination address but there was silence on the line and no data could be collected.

Action: Verify the destination address.

Not enough data
The Gateway dialed the destination address but the data collected was not enough to extract tone characteristics.

Action: Verify the destination address and ensure that the destination does not forward to another extension or that the destination is not answered.

Failed Validation unexpected CP tone

The Gateway dialed the destination address and was able to extract tone characteristics, but when the Gateway dialed the destination address to validate the newly learned tone, an unexpected call progress (CP) tone was detected. The detected CP tone is listed in the error message. This detected CP tone did not match the newly learned tone or any expected conflicts. This detected CP tone may be a tone that already exists in the tone definition table or another newly learned tone.

Action: Using the listed tone characteristics and the unexpected tone event that was detected, it may be possible to compare tones and determine which of the tones was actually detected. If the tone definitions only overlap in the time deviation, then it may be possible to manually add the tone and adjust the time deviation of the new tone and the conflicting tone to give two unique tones. If the two tones are the same, then only one tone can be used.

Failed Validation Timeout

The Gateway dialed the destination address and was able to extract tone characteristics, but when the Gateway dialed the destination address to validate the newly learned tone, no CP tone was detected.

Action: Validate the destination address.

Existing Tones

The tones listed in this section were successfully learned but found to match tones already in the configuration. Since these tones are already in the tone definition table no further action is needed.

Unique Tones

The tones listed in this section were successfully learned and validated without error or conflicts. If the “Add tone to Configuration?” checkbox is checked, these tones will be added to the tone definition table when you select the Apply Changes button.

Conflicting Tones

The tones listed in this section were successfully learned but were found to conflict with other tones. You can select only one of the conflicting tones to be added to or kept in the configuration. If the tone that conflicts with the selected tone is already in the tone definition table, it will be deleted when you select the Apply Changes button. If the conflicting tone is another newly learned tone, it will be ignored.

Note: If the conflicting tones do not overlap by much and it is desired to keep both conflicting tones, you can print the results page or write down the tone characteristics for the conflicting tone and add it to the tone definition table manually. You will need to manually adjust the tone characteristics of the conflicting tones to remove the conflict.

Learn Tone Issues and Possible Solutions

When comparing tone definitions, both the frequency and timing need to be considered.

To compare frequencies, both the frequency itself and the frequency deviation are needed. When a frequency is detected on the line, it is compared to the bandwidth of the tone definition. The

bandwidth is specified as all of the frequencies from the specified frequency minus the frequency deviation to the specified frequency plus the frequency deviation. For example, if the frequency is specified as 400 Hz and the frequency deviation is specified as 50 Hz then the bandwidth is 350 Hz to 450 Hz.

The frequencies of tone definitions are considered to match if their bandwidths overlap. For example, if tone 1 has frequency 400 Hz with frequency deviation of 50 Hz, and tone 2 has frequency 460 Hz with frequency deviation of 20 Hz, then these frequencies would match. The bandwidth of tone 1 is 350 through 450 Hz, and the bandwidth of tone 2 is 440 through 480 Hz.

Some common learn tone issues and possible solutions are provided here:

An On/Off tone is detected while validating a stutter dialtone

A stutter dial tone is a dial tone that has a short period of 2 or 3 cycles of On/Off cadence before the dialtone is on continuously. In most cases the stutter dial tone will not be an issue as the dial tone will have distinct frequencies. In those rare cases where the dial tone shares the same frequencies as On/Off tones, it may be possible that the dialtone will experience validation errors. This may happen if the dial tone stutter period matches the on and off timing of a tone with the same frequencies. Since there is no conflict between Continuous tones and On/Off tones, the conflict between On/Off tones and the stutter part of the stutter dial tone will not be detected automatically.

Possible solution: Adjust the timing of the On/Off tone, or set the number of cadence cycles for the On/Off tone to a value greater than 1.

An On/Off tone with double cadence conflicts with another On/Off tone

A double cadence is an On/Off cadence with two different on and two different off periods. For example, a ringback tone may have the following cadence: on for 500 msec, off for 500 msec, on for 1500 msec, off for 3500 msec. This cadence will automatically be learned as an On/Off tone with time on as 1000 msec with deviation of 500 msec and Off time of 2000 with deviation of 1500 msec. Since this timing has such wide deviation it may conflict with other tones with the same frequency. For example, if there is a busy tone with the same frequency and the cadence of on for 500 msec, off for 500 msec, then this busy will conflict with the ringback definition.

Possible solution: To prevent the busy tone event during ringback, increase the number of cadence cycles for the busy tone to 2. To prevent ringback event during busy, change the cadence type of the ringback to OnOffQuick. Set the first time value to a value that is greater than the time on for the busy tone, but less than the larger on time of the ringback. For this example, 1000 msec would be a good choice. Set the second time value to a value larger than the longest off time for the ringback. For this example, 3600 msec would be a good choice.

Note: Changing the cadence type to OnOffQuick may not work if there is already a continuous type or OnOffQuick type with matching frequencies. In this case, the Gateway will not be able to reliably detect the tone with a double cadence.

4.2.5 Validating Call Progress Tones

If the timing and frequency information is available and the desire is to only validate the information, then use the Validate Only button. The Media Gateway will dial a destination address where the call progress tone to be tested can be heard. The unit will then call each destination address and determine if the specified call progress tone is detected.

The validation process is useful when there was an error in learning new tones and some tone characteristics have been manually adjusted to remove conflicts. The Validate Only button can be used to validate the changes.

Follow these steps to validate the timing and frequency characteristics of call progress tones.

1. For each call progress tone that is to be validated, determine how that tone can be generated by the system.

2. Verify that the system will generate the call progress tone that is desired by dialing that destination address from an extension and listening to verify the tone.

3. Navigate to the Learn Tone Web page (see Figure 21) and enter the destination addresses for the tones to be learned.

4. Click on the Validate Only button.

5. Wait for the Validate Tone Results page to be displayed. A progress page will be displayed showing progress messages as the system proceeds.

4.2.6 Validate Tone Progress

Once the Validate Only button has been selected, the Media Gateway will start the process of validating the call progress tone. A progress page will automatically appear while the Gateway is validating the tones and progress messages will be posted.

The message, Validating, means that the Gateway is redialing the destination addresses and testing to see if the call progress tone definitions can detect the tones.

The Cancel button cancels the current session and returns to the Learn Tone Web page.

4.2.7 Validate Tone Results

When the Media Gateway has finished validating the tones, a results page will be displayed.

The validated tones are grouped into the following tone results:

- Validation Errors
- Validated Tones

After viewing the validation results, you can return to the Manual Tones Web Page (see Figure 20) by clicking on the Return to Tone Configuration link.

Validation Errors

An error occurred when trying to validate the tones listed in this section. These tones cannot be added to the configuration due to the error encountered. Possible reasons for error include:

Failed Validation unexpected CP tone

When the Gateway dialed the destination address to validate the tone, an unexpected call progress (CP) tone was detected. The detected CP tone is listed in the error message.

Failed Validation Timeout

When the Gateway dialed the destination address to validate the tone, no CP tone was detected.

Validated Tones

The tones listed in this section were successfully validated.

Routing Table 5

Information about Routing Table and how it is supported by the Dialogic® Media Gateway is described in the following sections:

- Routing Table Overview 193
- Router Configuration 195
- Offline Testing 217
- Call Routing Examples 220

5.1 Routing Table Overview

The Routing Table (previously referred to as the Dial Plan) describes a set of rules used to define the characteristics of a call routed through the gateway. It allows telephony requests to be processed if the inbound information is matched against configured set of rules and determines if there is an available outbound route for the request. The primary characteristics include the destination address and the CPID (call party identification) information. The Routing Table affects calls originating from the VoIP side, and calls originating from the TDM (T1, E1, analog, etc.) side.

5.1.1 VoIP to TDM Calls

Calls originating from the VoIP interface have two associated URLs. One contains the originating address and a calling number. The other contains a destination address and a called number. The Routing Table uses this information to determine a final TDM destination for the call. Let's say a call occurs as follows:

From: 101@172.16.3.131
To: 8675309@172.16.3.200

The user at address 172.16.3.131 is attempting to connect to the gateway at 172.16.3.200. The calling number is 101, and the called number is 8675309.

Figure 22 below shows how this information enters the Routing Table control.

Figure 22. VoIP to TDM calls

The Routing Table configuration will determine what calling and called numbers are sent to the TDM device, as well as the physical TDM destination.

5.1.2 TDM to VoIP Calls

Calls originating from the TDM interface have a physical source, along with Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name information. The Routing Table uses a combination of the physical source, Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name to determine the URL of the destination call. Take the case where a call occurs as follows on a T1 interface:

Physical Source: Interface 2, Channel 6
Calling Number: 5402
Called Number: 95551212

A call is being received on the 6th channel of physical interface 2. The calling number (the caller) is 5402, and the number called (dialed) is 95551212.

Figure 23 below shows how this information enters the Routing Table control.

Figure 23. TDM to VoIP calls

The Routing Table configuration will determine the calling number, called number, redirect number, and VoIP address of the destination call.

5.2 Router Configuration

The Router Configuration manages the configuration of the Routing Table for the gateway. The Routing Table defines how all telephony requests are routed through the gateway between networks. It also defines the call-party information manipulation rules for passing call-party information between networks. TDM Trunk Groups and VoIP Host Groups should be created first, as they are referenced by both the Inbound TDM and Inbound VoIP routing rules.

The Web interface is used for configuring all 4 tables of the Routing Table. New rules can be added, existing rules can be modified, and existing rules can be deleted.

Note: Changes will not be permanent until the "Submit" button is clicked.

5.2.1 Determining the Call Destination

The Routing Table uses four tables configured by the user to determine the final destination.

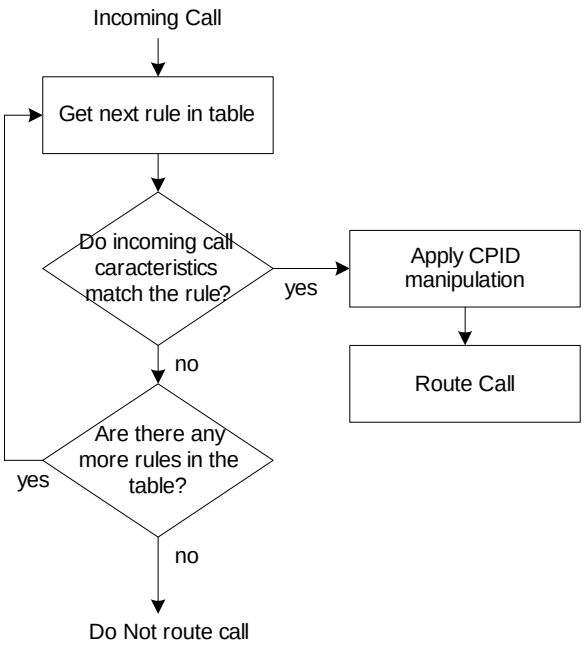
- The Inbound TDM Rules Table provides a set of rules used to determine the route of incoming TDM calls.
- The Inbound VoIP Rules Table provides a set of rules used to determine the route of incoming VoIP calls.
- The TDM Trunk Groups Table is used to divide the TDM ports and channels into groups that can be used for matching inbound TDM calls or routing outbound TDM calls.
- The VoIP Host Groups Table is used to divide the VoIP endpoint IP Addresses into groups that can be used for routing outbound VoIP calls.

  Note: The VoIP Host Groups Table is not used for Inbound VoIP call matching.

As a call enters the gateway, the incoming characteristics of the call are matched against the Routing Table rules in the appropriate Routing Table from top down. If a match is found, the CPID information is updated based on the CPID manipulation entry in the table assigned to the matched rule. The call is then sent to the destination assigned in the table.

Figure 24 below shows call routing flow for Routing Table.

Figure 24. Routing Table Call Routing Flow



If no match is found, the call is not routed by the Routing Table.

5.2.2 Inbound TDM Rules Table

The Inbound TDM Rules is the main table used to route calls originating from the TDM interface. When a call is received, the incoming call characteristics are compared against the values entered in the first row of the table. If a match is found, an action is taken. If no match is found, the next row in the table is checked. Rows will be tested until a match is found, or there are no more entries in the table.

For a test to pass, the channel of the incoming call must reside in the specified channel pool. The Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name must also match the rule entered in the table. Once a match occurs, the CPID will be modified based on the CPID manipulation rule selected in the table. Next, the outgoing call will be made to the VoIP address specified in this table.

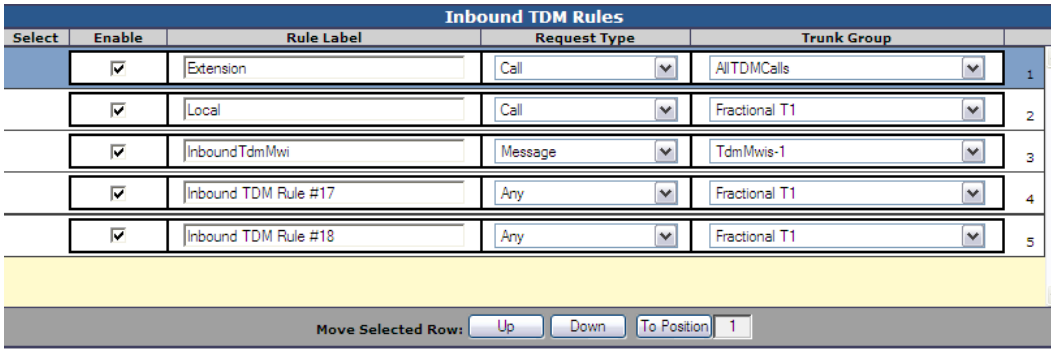
If no rows in the table successfully match the incoming call, the call is not routed.

5.2.2.1 Inbound TDM Rules

Use this table to define the routing rules for all telephony requests that are inbound from the TDM network. Rules are evaluated from the top down until a matching rule is found.

Figure 25 is a screen shot of the Inbound TDM Rules configuration Web page:

Figure 25. Inbound TDM Rules Configuration Web Page



Select to configure the routing rules for requests that are inbound from the TDM network.

Select - Click anywhere in this column to select an entry. When a rule is highlighted, the associated configuration is shown in the tables below. The selected row may be moved up or down in the table by dragging and dropping it into a different position in the list, or deleted. The selected row may also be moved by clicking on the Move Selected Row buttons: Up, Down, or To Position.

Note: It is a common error to modify the configuration of the incorrect rule. Make sure that the correct rule is selected before modifying the configuration.

Enable - Check this box to enable the rule. If this box is not checked, this rule is not checked for incoming calls.

Rule Label - Specifies the label for this entry. This label is used only to help identify the rule to the administrator. It is not used by any other tables in the Routing Table configuration.

Request Type - Specifies the type of telephony request to which the rule applies:

- Any = All telephony request types.
- Call = Calls only.
- Message = Message-waiting notifications only.

Trunk Group - Specifies the name of the TDM Trunk Group on which the request must arrive at the gateway in order to match the rule.

5.2.2.2 Inbound TDM Request Matching

Defines the call-party information expressions that are used to match an inbound telephony request from the TDM network to this routing rule. The Inbound TDM Request Matching is performed when the inbound TDM request matches the Request Type and Trunk Group of one of the defined rules.

CPID Matching

Part of the criteria for matching the characteristics of an incoming call to a rule in the Routing Table is number matching. The incoming Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name are tested against a match rule defined in the Routing Table. If the match fails, this routing rule table entry will fail.

Table 10 shows the syntax used for CPID Matching.

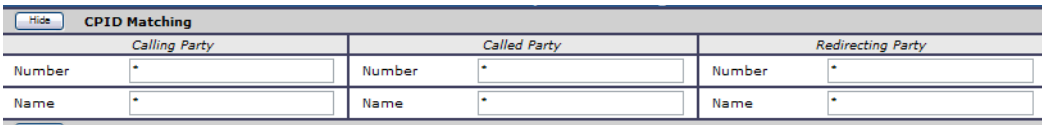
Table 10. Syntax Used for CPID Matching

Number Matching	
Token	**Description**
*	Matches any number string when entered alone.
0 1 2 3 4 5 6 7 8 9	Identifies a specific digit.
[digit_string-digit_string]	Specifies a range of digit strings. Identifies any digit string that is included in the range. The range delimiters must both contain the same number of digits. A digit string must be no more than 9 digits in length.
[digit_string-digit_string, digit_string]	Specifies a range of digit strings as a comma separated list. The range delimiters and comma separated number strings must all contain the same number of digits.
x	Matches any *single* digit.
.	Matches an arbitrary number of digits at the end of a dial string.
,	Creates a compound expression.
[blank]	Matches a blank (or missing) entry.

Name Matching	
Token	**Description**
*	Matches any number string when entered alone.
name	Must match the specified name exactly.

Figure 26 is a screen shot of the CPID Matching configuration Web page:

Figure 26. CPID Matching Configuration Web Page



Calling Party Number - The formula as specified in the section above that is used to match the calling number of the incoming call.

Calling Party Name- The formula as specified in the section above that is used to match the calling name of the incoming call.

Called Party Number - The formula as specified in the section above that is used to match the called number of the incoming call.

Called Party Name - The formula as specified in the section above that is used to match the called name of the incoming call.

Redirecting Party Number - The formula as specified in the section above that is used to match the redirect number of the incoming call.

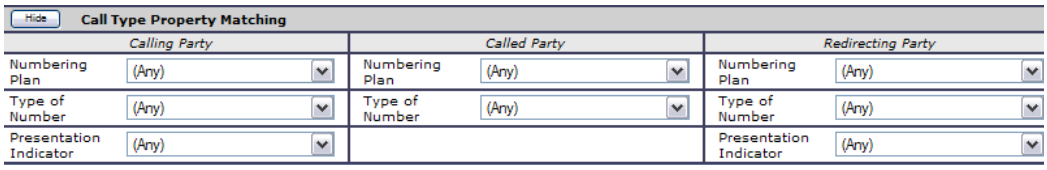
Redirecting Party Name - The formula as specified in the section above that is used to match the redirect name of the incoming call.

Call Type Property Matching

Specifies matching of incoming ISDN Call Type characteristics. On a TDM interface using the ISDN protocol, incoming calls may have Call Type information associated with them. Call Type Property Matching allows calls to be matched based on this information. Call Type has three properties and can apply to the Calling, Called, or Redirecting party. Any, all, or none of the properties may exist in an ISDN call.

Figure 27 is a screen shot of the Call Type Property Matching configuration Web page:

Figure 27. Call Type Property Matching Configuration Web Page



Calling Party Numbering Plan - Numbering Plan match for Calling Number (ISDN only).

- Any
- Unknown
- ISDN_E.164
- Data_X.121
- Telex_F.69
- Standard
- Private
- Reserved

Calling Party Type of Number - Type of Number match for Calling Number (ISDN only).

- Any
- Unknown
- International
- National
- Network
- Subscriber
- Abbreviated
- Reserved
- Level2Regional
- Level1Regional
- PISNSpecific
- Level0Regional

Calling Party Presentation Indicator - Presentation Indicator match for Calling Number (ISDN only).

- Any
- Allowed
- Restricted
- NotAvail
- Reserved

Called Party Numbering Plan - Numbering Plan match for Called Number (ISDN only).

- Any
- Unknown
- ISDN_E.164
- Data_X.121
- Telex_F.69
- Standard
- Private
- Reserved

Called Party Type of Number - Type of Number match for Called Number (ISDN only).

- Any
- Unknown
- International
- National
- Network
- Subscriber
- Abbreviated
- Reserved
- Level2Regional
- Level1Regional
- PISNSpecific
- Level0Regional

Redirecting Party Numbering Plan - Numbering Plan match for Redirect Number (ISDN only).

- Any
- Unknown
- ISDN_E.164
- Data_X.121
- Telex_F.69
- Standard
- Private
- Reserved

Redirecting Party Type of Number - Type of Number match for Redirect Number (ISDN only).

- Any
- Unknown
- International
- National
- Network
- Subscriber
- Abbreviated
- Reserved
- Level2Regional
- Level1Regional
- PISNSpecific
- Level0Regional

Redirecting Party Presentation Indicator - Presentation Indicator match for Redirect Number (ISDN only).

- Any
- Allowed
- Restricted
- NotAvail
- Reserved

5.2.2.3 Outbound Routes

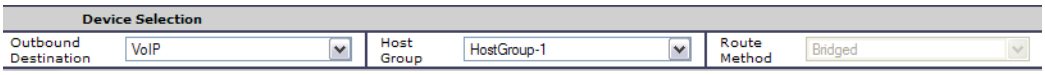
Defines how the request is to be routed by the gateway if the TDM inbound request matches the Request Type, Trunk Group, and CPID matching of a given rule in the Inbound TDM Rules table. This defines the network to which the request is to be routed as well as the CPID Manipulation expressions that are to be used to manipulate CPID prior to routing the request.

Device Selection

Specifies the destination network of the route and how the request is to be routed.

Figure 28 is a screen shot of the Device Selection configuration Web page:

Figure 28. Device Selection Configuration Web Page



Outbound Destination - Specifies the network to which the request is to be routed:

- VoIP = Route to the VoIP network.
- TDM = Route to the TDM network.
- Blocked = Request is not routed. The inbound request will be immediately rejected.

Host Group - Name of VoIP Host Group to which the request is to be routed.

Note: This selection box is only present when the Outbound Destination is set to VoIP.

Trunk Group - Specifies the name of TDM Trunk Group to which the request is to be routed.

Note: This selection box is only present when the Outbound Destination is set to TDM.

Route Method - Specifies the method used for routing requests:

- Redirect = Requests are redirected to the destination, which means that the gateway does not bridge the two networks. Valid only when routing from/to the same network (ex. VoIP->VoIP, TDM->TDM).
- Bridged = Requests are bridged through the gateway between the inbound and outbound network.

Note: This selection box is only present when the Outbound Destination is set to TDM.

CPID Manipulation

CPID Manipulation defines rules for how Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name appear in the destination call. Typically, the calling and called numbers from the incoming call are simply passed through to the outgoing call. However, there are cases where it is desirable to alter either or both of these numbers. A CPID Manipulation rule is applied to all calls routed via the Routing Table. Each of the rules applies a formula using the incoming CPID information as inputs to determine the outgoing call's CPID.

Table 11 shows the syntax used for CPID Manipulation.

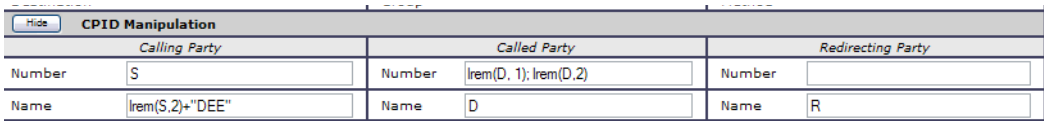
Table 11. Syntax Used for CPID Manipulation

Number Manipulation			
Rule Syntax	**Description**	**Example**	**Example Result**
S	Source (calling) number	S	7168675309
D	Destination (called) number	D	5551212
R	Redirection number	R	4815162342
S#	Source (calling) number	S#	7168675309
D#	Destination (called) number	D#	5551212
R#	Redirection number	R#	4815162342
S@	Source (calling) name	S@	Amanda
D@	Destination (called) name	D@	Sydney
R@	Redirection name	R@	Dave
“ ”	Takes what’s in quotes as literal	“353”	353
+	Concatenate	“800” + D	8005551212
lext(str, n)	Extract ***n*** characters from left of ***str***	lext(S, 3)	716
rext(str, n)	Extract ***n*** characters from right of ***str***	rext(S, 4)	5309
lrem(str, n)	Remove ***n*** characters from left of ***str***	lrem(S@, 3)	nda
rrem(str, n)	Remove ***n*** characters from right of ***str***	rrem(D, 4)	555
mext(str, pos, n)	Extract ***n*** characters from ***str*** starting ***pos*** digits from left	mext(S, 5, 2)	53
repl(str, old, new)	Find 1st occurrence of ***old*** in ***str*** and replace with ***new***	repl(D, “12”, “46”)	5554612
	Mix multiple rules	S@ + “-at-” + rext(S,3)	Amanda-at-309

Name Manipulation			
Rule Syntax	**Description**	**Example**	**Example Result**
S	Source (calling) number	S	Amanda
D	Destination (called) number	D	Sydney
R	Redirection number	R	Dave
S#	Source (calling) number	S#	7168675309
D#	Destination (called) number	D#	5551212
R#	Redirection number	R#	4815162342
S@	Source (calling) name	S@	Amanda
D@	Destination (called) name	D@	Sydney
R@	Redirection name	R@	Dave
“ ”	Takes what’s in quotes as literal	“353”	353
+	Concatenate	“800” + D	800Sydney
lext(str, n)	Extract ***n*** characters from left of ***str***	lext(S, 3)	Ama
rext(str, n)	Extract ***n*** characters from right of ***str***	rext(S@, 4)	anda
lrem(str, n)	Remove ***n*** characters from left of ***str***	lrem(S#, 3)	8675309
rrem(str, n)	Remove ***n*** characters from right of ***str***	rrem(D#, 4)	555
mext(str, pos, n)	Extract ***n*** characters from ***str*** starting ***pos*** digits from left	mext(S, 1, 3)	man
repl(str, old, new)	Find 1st occurrence of ***old*** in ***str*** and replace with ***new***	repl(D, “ydne”, “all”)	Sally
	Mix multiple rules	S@ + “-at-” + rext(S#,3)	Amanda-at-309

Figure 29 is a screen shot of the CPID Manipulation configuration Web page:

Figure 29. CPID Manipulation Configuration Web Page



Calling Party Number - The formula as specified in the section above that determines the calling number of the outgoing call.

Calling Party Name- The formula as specified in the section above that determines the calling name of the outgoing call.

Called Party Number - The formula as specified in the section above that determines the called number of the outgoing call.

Called Party Name - The formula as specified in the section above that determines the called name of the outgoing call.

Redirecting Party Number - The formula as specified in the section above that determines the redirect number of the outgoing call.

Redirecting Party Name - The formula as specified in the section above that determines the redirect name of the outgoing call.

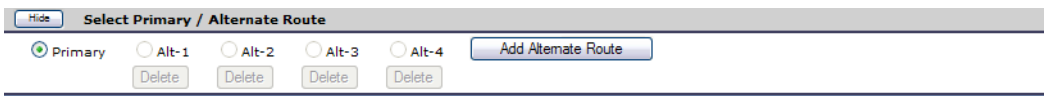
Select Primary / Alternate Route

One Primary and multiple Alternate routes may be defined for a given routing rule if the gateway is unable to route the request to the Primary. The Alternate routes will then be attempted until the request is successfully routed, or if there are no more Alternate routes to try. Examples of failure that may occur before the call is connected to the outbound destination include: for SIP, any 4xx, 5xx or 6xx response; for TDM, error tone, congestion tone, special information tone; for any, the user is busy, the physical interface is unavailable.

Note: The Primary and Alternate routes differ only in the Outbound Routes section. All other parts of the rule (CPID matching, etc.) are identical between the primary and alternate routes.

Figure 30 is a screen shot of the Select Primary / Alternate Route configuration Web page:

Figure 30. Select Primary / Alternate Route Configuration Web Page



5.2.3 Inbound VoIP Rules Table

The Inbound VoIP Rules is the main table used to route calls originating from the VoIP interface. When a call is received, the incoming call characteristics are compared against the values entered in the first row of the table. If a match is found, an action is taken. If no match is found, the next row in the table is checked. Rows will be tested until a match is found, or there are no more entries in the table.

The VoIP address, Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name **all** need to match the incoming call for the test to pass. Once a match occurs, the CPID will be modified based on the CPID manipulation rule selected in the table. Next, the call will be routed according to the matched rules associated Outbound Routes fields.

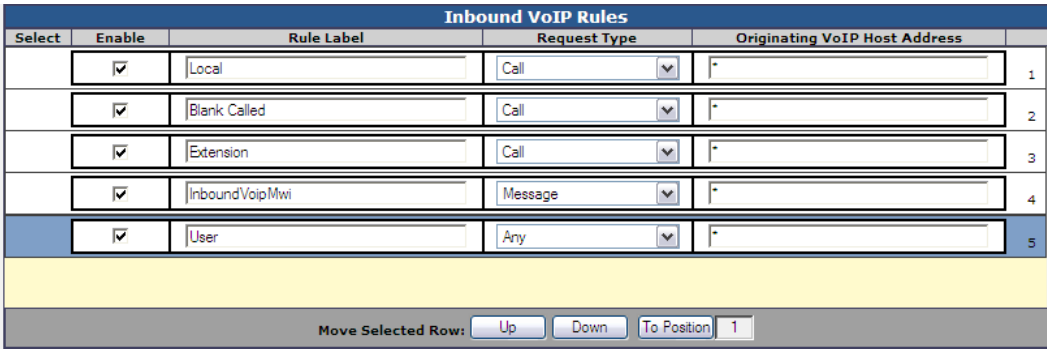
If no rows in the table successfully match the incoming call, the call is not routed.

5.2.3.1 Inbound VoIP Rules

Use this table to define the routing rules for all telephony requests that are inbound from the VoIP network. Rules are evaluated from the top down until a matching rule is found.

Figure 31 is a screen shot of the Inbound VoIP Rules configuration Web page:

Figure 31. Inbound VoIP Rules Configuration Web Page



Select to configure the routing rules for requests that are inbound from the VoIP network.

Select - Click anywhere in this column to select an entry. When a rule is highlighted, the associated configuration is shown in the tables below. The selected row may be moved up or down in the table by dragging and dropping it into a different position in the list, or deleted. The selected row may also be moved by clicking on the Move Selected Row buttons: Up, Down, or To Position.

Note: It is a common error to modify the configuration of the incorrect rule. Make sure that the correct rule is selected before modifying the configuration.

Enable - Check this box to enable the rule. If this box is not checked, this rule is not checked for incoming calls.

Rule Label - Specifies label for this entry. This label is used only to help identify the rule to the administrator. It is not used by any other tables in the Routing Table configuration.

Request Type - Specifies the type of telephony request to which the rule applies:

- Any = All telephony request types.
- Call = Calls only.
- Message = Message-waiting notifications only.

Originating VoIP Host Address - Specifies the VoIP hosts from which requests are accepted for use by the rule. The originating VoIP host's address must match this expression in order for the telephony request to match the rule.

Table 12 shows the syntax for VoIP Host Address.

Table 12. Syntax for VoIP Host Address

Token	Description
*	Matches any host address.
address	Address may be a FQDN or an IP address, and must match exactly.
IP_Addr/N	Matches any IP address whose left-most N bits match the pattern in IP_Addr. ex: 10.10.11.0/24 would match any IP address of the form 10.10.11.XX, acting the same as a subnet mask of 255.255.255.0.
Note: A single entry may contain multiple hostnames and/or IP_Addr, each separated by a comma ','. For example: **10.10.11.12,10.10.11.13,machine.domain.com**.	

5.2.3.2 Inbound VoIP Request Matching

Defines the call-party information expressions that are used to match an inbound telephony request from the VoIP network to this routing rule. The Inbound VoIP Request Matching is performed when the inbound VoIP request matches the Request Type and Originating VoIP Address of one of the defined rules.

CPID Matching

Part of the criteria for matching the characteristics of an incoming call to a rule in the Routing Table is number matching. The incoming Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name is tested against a match rule defined in the Routing Table. If the match fails, this routing rule table entry will fail.

Table 13 shows the syntax used for CPID Matching.

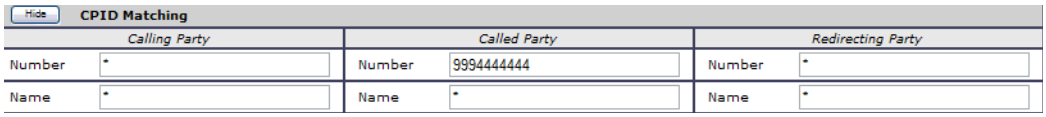
Table 13. Syntax Used for CPID Matching

Number Matching	
Token	**Description**
*	Matches any number string when entered alone.
0 1 2 3 4 5 6 7 8 9	Identifies a specific digit.
[digit_string-digit_string]	Specifies a range of digit strings. Identifies any digit string that is included in the range. The range delimiters must both contain the same number of digits. A digit string must be no more than 9 digits in length.
[digit_string-digit_string, digit_string]	Specifies a range of digit strings as a comma separated list. The range delimiters and comma separated number strings must all contain the same number of digits.
x	Matches any *single* digit.
.	Matches an arbitrary number of digits at the end of a dial string.
,	Creates a compound expression.
[blank]	Matches a blank (or missing) entry.

Name Matching	
Token	**Description**
*	Matches any number string when entered alone.
name	Must match the specified name exactly.

Figure 32 is a screen shot of the CPID Matching configuration Web page:

Figure 32. CPID Matching Configuration Web Page



Calling Party Number - The formula as specified in the section above that is used to match the calling number of the incoming call.

Calling Party Name- The formula as specified in the section above that is used to match the calling name of the incoming call.

Called Party Number - The formula as specified in the section above that is used to match the called number of the incoming call.

Called Party Name - The formula as specified in the section above that is used to match the called name of the incoming call.

Redirecting Party Number - The formula as specified in the section above that is used to match the redirect number of the incoming call.

Redirect Party Name - The formula as specified in the section above that is used to match the redirect name of the incoming call.

5.2.3.3 Outbound Routes

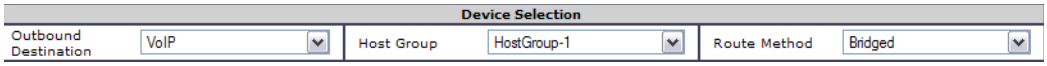
Defines how the request is to be routed by the gateway. This defines the network to which the request is to be routed as well as the CPID Manipulation expressions that are to be used to manipulate CPID prior to routing the request.

Device Selection

Specifies the destination network of the route and how the request is to be routed.

Figure 33 is a screen shot of the Device Selection configuration Web page:

Figure 33. Device Selection Configuration Web Page



Outbound Destination - Specifies the network to which the request is to be routed:

- VoIP = Route to the VoIP network.
- TDM = Route to the TDM network.
- Blocked = Request is not routed. The inbound request will be immediately rejected.

Host Group - Name of VoIP Host Group to which the request is to be routed.

Note: This selection box is only present when the Outbound Destination is set to VoIP.

Trunk Group - Name of TDM Trunk Group to which the request is to be routed.

Note: This selection box is only present when the Outbound Destination is set to TDM.

Route Method - Method used for routing requests:

- Redirect = Request is redirected to the destination. The gateway does not bridge the two networks. This is valid only when routing from/to the same network.
- Bridged = Request is bridged through the gateway between the inbound and outbound network.

Note: This selection box is only supported on DMG2000. If the Outbound Destination is VoIP on DMG1000, then the Route Method is forced to Redirect.

CPID Manipulation

CPID Manipulation defines rules for how Calling Number, Calling Name, Called Number, Called Name, Redirect Number, and Redirect Name appear in the destination call. Typically, the calling and called numbers from the incoming call are simply passed through to the outgoing call. However, there are cases where it is desirable to alter either or both of these numbers. A "CPID Manipulation" rule is applied to all calls routed via the Routing Table. Each of the rules applies a formula using the incoming CPID information as inputs to determine the outgoing call's CPID.

Table 14 shows the syntax used for CPID Manipulation.

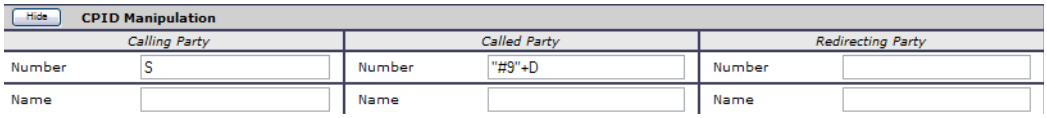
Table 14. Syntax Used for CPID Manipulation

Number Manipulation			
Rule Syntax	**Description**	**Example**	**Example Result**
S	Source (calling) number	S	7168675309
D	Destination (called) number	D	5551212
R	Redirection number	R	4815162342
S#	Source (calling) number	S#	7168675309
D#	Destination (called) number	D#	5551212
R#	Redirection number	R#	4815162342
S@	Source (calling) name	S@	Amanda
D@	Destination (called) name	D@	Sydney
R@	Redirection name	R@	Dave
" "	Takes what's in quotes as literal	"353"	353
+	Concatenate	"800" + D	8005551212
lext(str, n)	Extract ***n*** characters from left of ***str***	lext(S, 3)	716
rext(str, n)	Extract ***n*** characters from right of ***str***	rext(S, 4)	5309
lrem(str, n)	Remove ***n*** characters from left of ***str***	lrem(S@, 3)	nda
rrem(str, n)	Remove ***n*** characters from right of ***str***	rrem(D, 4)	555
mext(str, pos, n)	Extract ***n*** characters from ***str*** starting ***pos*** digits from left	mext(S, 5, 2)	53
repl(str, old, new)	Find 1st occurrence of ***old*** in ***str*** and replace with ***new***	repl(D, "12", "46")	5554612
	Mix multiple rules	S@ + "-at-" + rext(S,3)	Amanda-at-309

Name Manipulation			
Rule Syntax	**Description**	**Example**	**Example Result**
S	Source (calling) number	S	Amanda
D	Destination (called) number	D	Sydney
R	Redirection number	R	Dave
S#	Source (calling) number	S#	7168675309
D#	Destination (called) number	D#	5551212
R#	Redirection number	R#	4815162342
S@	Source (calling) name	S@	Amanda
D@	Destination (called) name	D@	Sydney
R@	Redirection name	R@	Dave
“ ”	Takes what’s in quotes as literal	“353”	353
+	Concatenate	“800” + D	800Sydney
lext(str, n)	Extract ***n*** characters from left of ***str***	lext(S, 3)	Ama
rext(str, n)	Extract ***n*** characters from right of ***str***	rext(S@, 4)	anda
lrem(str, n)	Remove ***n*** characters from left of ***str***	lrem(S#, 3)	8675309
rrem(str, n)	Remove ***n*** characters from right of ***str***	rrem(D#, 4)	555
mext(str, pos, n)	Extract ***n*** characters from ***str*** starting ***pos*** digits from left	mext(S, 1, 3)	man
repl(str, old, new)	Find 1st occurrence of ***old*** in ***str*** and replace with ***new***	repl(D, “ydne”, “all”)	Sally
	Mix multiple rules	S@ + “-at-” + rext(S#,3)	Amanda-at-309

Figure 34 is a screen shot of the CPID Manipulation configuration Web page:

Figure 34. CPID Manipulation Configuration Web Page



Calling Party Number - The formula as specified in the section above that determines the calling number of the outgoing call.

Calling Party Name- The formula as specified in the section above that determines the calling name of the outgoing call.

Called Party Number - The formula as specified in the section above that determines the called number of the outgoing call.

Called Party Name - The formula as specified in the section above that determines the called name of the outgoing call.

Redirecting Party Number - The formula as specified in the section above that determines the redirect number of the outgoing call.

Redirect Party Name - The formula as specified in the section above that determines the redirect name of the outgoing call.

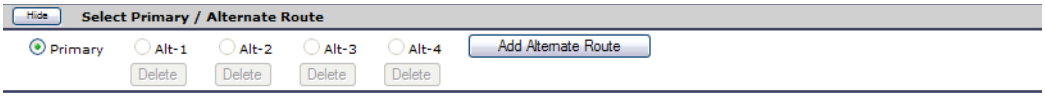
Select Primary / Alternate Route

One Primary and multiple Alternate routes may be defined for a given routing rule if the gateway is unable to route the request to the Primary. The Alternate routes will then be attempted until the request is successfully routed, or if there are no more Alternate routes to try. Examples of failure that may occur before the call is connected to the outbound destination include: for SIP, any 4xx, 5xx or 6xx response; for TDM, error tone, congestion tone, special information tone; for any, the user is busy, the physical interface is unavailable.

Note: The Primary and Alternate routes differ only in the Outbound Routes section. All other parts of the rule (CPID matching, etc.) are identical between the primary and alternate routes.

Figure 35 is a screen shot of the Select Primary / Alternate Route configuration Web page:

Figure 35. Select Primary / Alternate Route Configuration Web Page



5.2.4 TDM Trunk Groups Table

TDM Trunk Groups are logical groupings of ports and channels that define the physical port for the TDM side of the call. The port is a physical connection to the gateway (i.e. ISDN span, Analog line), while the channel is a logical connection within the port (i.e. T1 CAS line includes up to 24 channels while an Analog line has a single channel). All incoming TDM calls arrive on a physical port and a channel. TDM Trunk Groups are a way of identifying the port / channel combination. For some calls, it may be necessary to select the specific port and channel from a group of possible options. TDM Trunk Groups define how this is selected.

The configuration of TDM Trunk Groups are used for the routing of requests from/to the TDM network. TDM Trunk Groups are referenced by Inbound TDM Rules and Inbound VoIP Rules. TDM Trunk Groups are used to match inbound requests with routing rules and to specify outbound routes to the TDM network.

5.2.4.1 TDM Port Types

The gateway supports multiple types of TDM ports. The number of physical ports and channels supported vary.

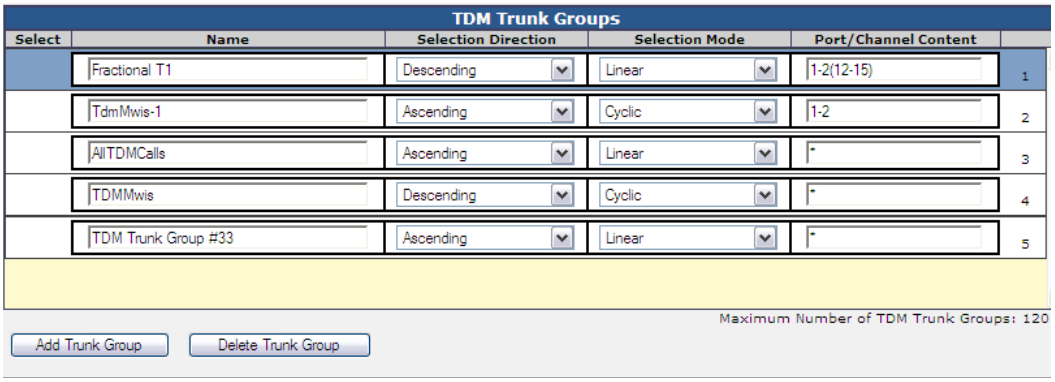
Table 15 shows the possibilities:

Table 15. TDM Port Types

Gateway Type	Maximum number of supported ports	Number of supported channels per port
Analog	8	1
Digital (PBX)	8	1
T1 - CAS	4	24
T1 - ISDN	4	23
E1	4	30

Figure 36 is a sample screen shot of the TDM Trunk Groups configuration Web page:

Figure 36. TDM Trunk Groups Configuration Web Page



Name - Defines the name of the trunk group. This name is referenced by routing rules that use the trunk group. Each group must have a unique name.

Selection Direction - Specifies the method of channel selection when selecting channels from the trunk group for outbound requests:

- Descending = Channels are selected from highest channel number to lowest.
- Ascending = Channels are selected from lowest channel number to highest.

Selection Mode - Specifies the method of channel selection when selecting channels from the trunk group for outbound requests:

Linear

- For gateways that have multiple channels per port, the highest free channel on the highest port is used for the next call (Descending) or the lowest free channel on the lowest port is used for the next call (Ascending).
- For gateways that have a single channel per port, the highest free channel is used for the next call (Descending) or the lowest free channel is used for the next call (Ascending).

Cyclic

- For gateways that have multiple channels per port:
  - For Descending, within a port the channels are cycled highest to lowest. After using the lowest channel, the next call is set to the next lower port's highest channel. When reaching the lowest channel on the lowest port, everything wraps.
  - For Ascending, within a port the channels are cycled lowest to highest. After using the highest channel, the next call is sent to the next higher port's lowest channel. When reaching the highest channel on the highest port, everything wraps.
- For gateways that have a single channel per port:
  - For Descending, the ports are selected in a round-robin fashion, highest to lowest (with wrapping).
  - For Ascending, the ports are selected in a round-robin fashion, lowest to highest (with wrapping).

Port/Channel Content - Each entry specifies the ports and channels that comprise the trunk group. The following defines how the trunk group contents are specified:

- 1. A digit specified outside of parenthesis is interpreted as a port.
- 2. A digit within parenthesis is interpreted as a channel.
- 3. Both ports and channels are 1-based.
- 4. Ranges are allowed, and specified using a dash '-' for ports and channels but does not allow combining a channel-range with a port-range (i.e. If specifying a port-range, channel-ranges are not allowed).
- 5. A comma is used to indicate another range, channel, or port for any given trunk group.
- 6. An entry of just the '*' character represents 'all ports and channels'.
- 7. Whitespace is ignored.

Examples:

* = All channels in all ports.

1-4 = All channels in ports 1 through 4.

1,3 = All channels in ports 1 and 3.

1(1-16),2(16-30),4 = Channels 1 through 16 on port 1 and channels 16 through 30 on port 2 and all channels on port 4.

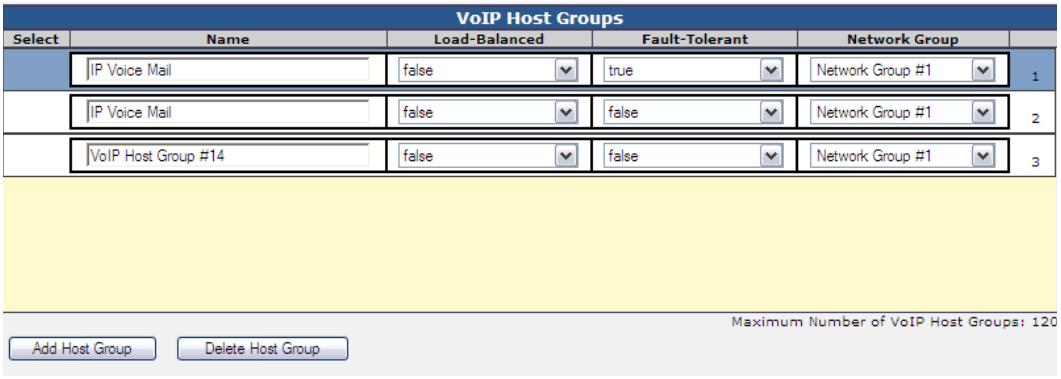
3(2,4-16) = Channel 2 and channels 4 through 16 on port 3.

5.2.5 VoIP Host Groups Table

The configuration of VoIP Host Groups are used for routing requests to the VoIP network. VoIP Host Groups are used to specify outbound routes to the VoIP network.

Figure 37 is a screen shot of the VoIP Host Groups configuration Web page:

Figure 37. VoIP Host Groups Configuration Web Page



Name - Defines the name of the host group. This name is referenced by routing rules that use the VoIP host group.

Load-Balanced - Enables/Disables load-balancing of outbound requests to the VoIP network. If 'true', then the Gateway will route an outbound VoIP request (call or MWI) to the next VoIP host in the group, in a round-robin fashion. If 'false', then the Gateway will not load-balance requests across multiple VoIP hosts in the group and will instead initially route all calls to the first VoIP host in the list.

Fault-Tolerant - Enables/Disables fault-tolerant handling of outbound VoIP requests. If 'true', then the Gateway will failover to the next configured VoIP host if an outbound VoIP route attempt fails. If 'false', then a failed outbound VoIP route attempt will not be retried.

Network Group - Select the VoIP Network Group configuration to be used for outbound VoIP requests.

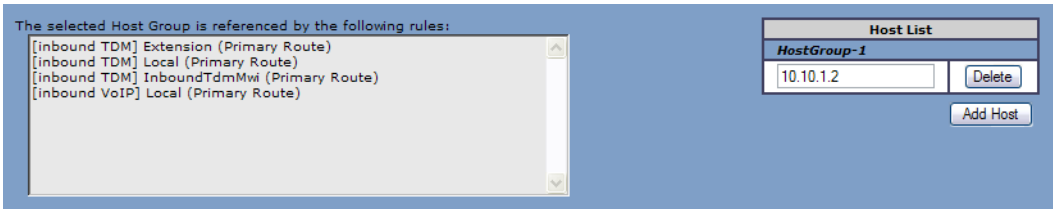
Note: The maximum number of supported VoIP Host Groups is 120 (DMG2000 only).

5.2.5.1 Host List

Use this table to Add hosts to this group and Delete hosts from this group.

Figure 38 is a screen shot of the Host List configuration Web page:

Figure 38. Host List Configuration Web Page



5.3 Offline Testing

An offline facility exists to test the Routing Table without receiving actual calls. To use this feature, navigate to the *Tests-->Router* link via the Media Gateway's user interface.

Two tables exist in this page. The "Inbound Route" table is used to take simulated call characteristics from an incoming call. First, the call direction is selected, and then the call information is entered. The "Outbound Route" table shows the result that would occur if the simulated incoming call passed through the Routing Table.

The fields for simulated data match the Routing Table exactly with one exception. On inbound TDM calls, enter the interface and channel instead of the TDM Trunk Group label. This helps test the TDM Trunk Group configuration as well as the routing configuration.

Figure 39. Inbound VOIP Route and Outbound Route

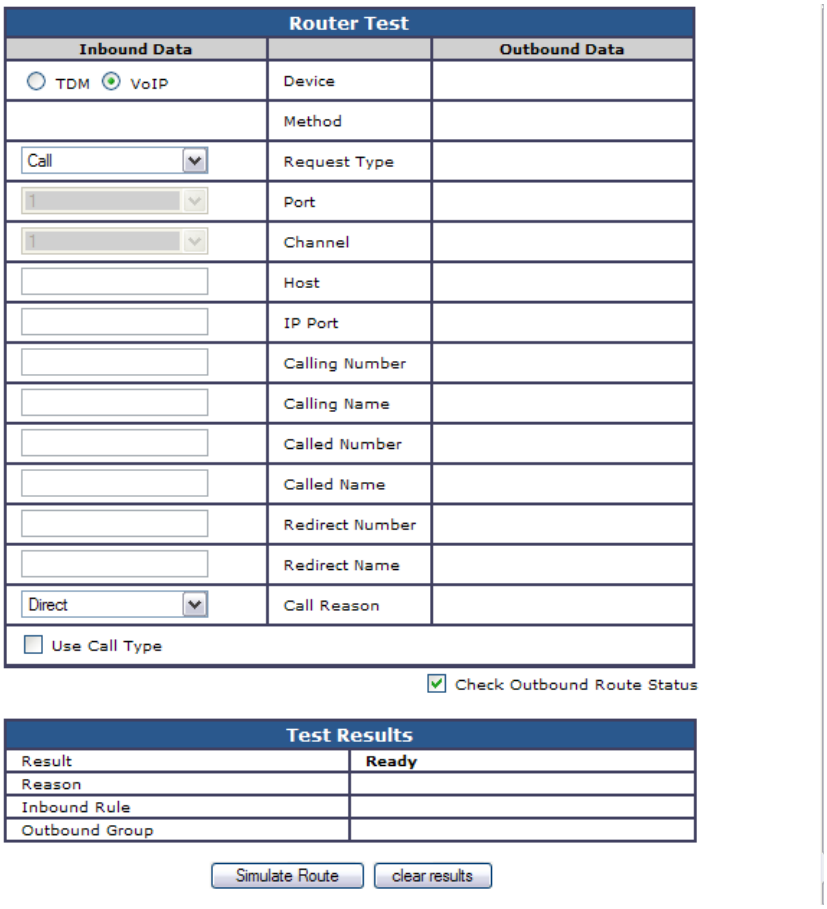
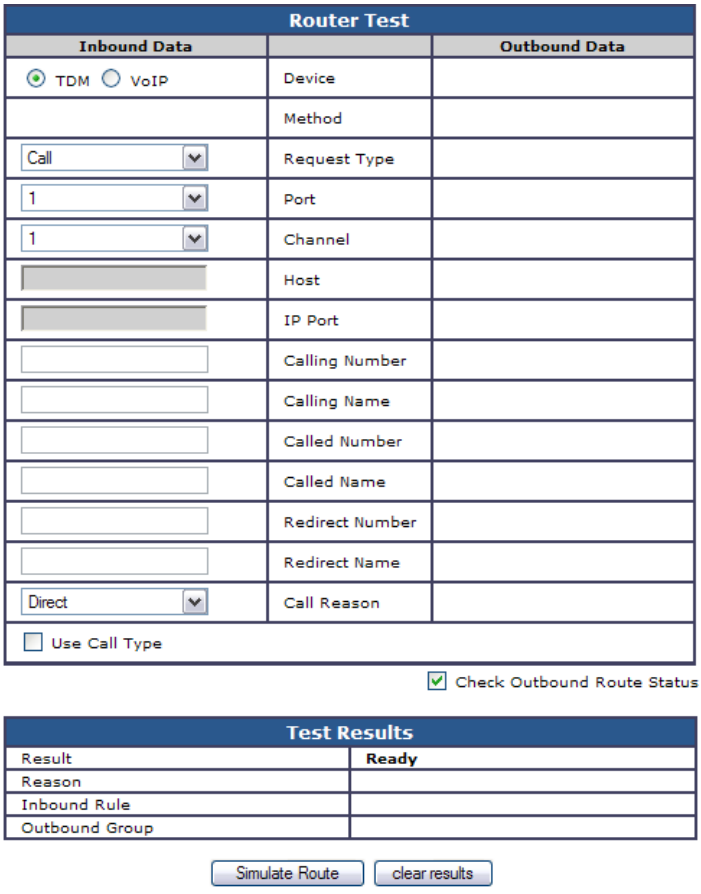


Figure 40. Inbound TDM Route and Outbound Route



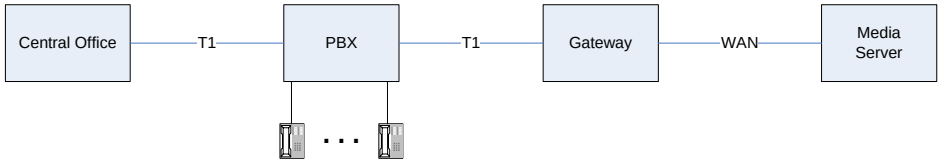
5.4 Call Routing Examples

Each example presents the requirements (i.e. how you want the calls routed) and the steps required to setup the routing table to meet the requirements. The examples are presented in increasing complexity and build on each other. The explanations presented in one example are not repeated in subsequent ones.

- Example #1 - Basic TDM to and from VoIP
- Example #2 - Basic Call with Load Balancing
- Example #3 - Basic Call with CPID Manipulation
- Example #4 - Basic Call with MWI
- Example #5 - Basic Call with Proxy
- Example #6 - Toll Bypass

Example #1 – Basic TDM to/from VoIP

System Configuration:



Requirements:

1. All inbound TDM calls must be routed to the media server (IP address 172.13.4.3).
2. All inbound VoIP calls must be routed out the TDM. Any available TDM channel can be used.

Steps:

1. Determine the required TDM Trunk Groups and the VoIP Host Groups.
   - From requirement #1 we have a TDM trunk group of "all TDM requests" and a VoIP host group of "172.13.4.3".
   - From requirement #2 we have a TDM trunk group of "any TDM channel". There is no corresponding "All VoIP Requests" host group since VoIP host groups are not used for inbound VoIP matching.
2. Set up TDM Trunk Groups.

Name	Selection Direction	Selection Mode	Port/Channel Content
all TDM requests (1)	(2)	(2)	* (3)
any TDM channel	Ascending (4)	Cyclic (4)	*

Notes:

1. The name of the group is your choice but it must be unique within the trunk group set.
2. Selection Direction and Selection Mode are only used for outbound TDM requests. Since this is an inbound group they are don't care.
3. An asterisk specifies all ports and channels.
4. If there are problems, **Cyclic** and **Ascending** are fairly easy to debug. So use them unless there are overriding site requirements.

3. Set up VoIP Host Groups.

Name	Load Balanced	Fault Tolerant	Host Summary
All to 172.13.4.3	false (1)	false (1)	172.13.4.3

Note: Load Balanced and Fault Tolerant are advanced concepts and are presented in subsequent examples. For now set to false.

4. Set up Inbound TDM Rules.

Inbound TDM Rules			
Enable	Rule Label	Request Type	Trunk Group
X	**All Inbound TDM**	**Any**	**All TDM Requests**

Inbound TDM Request Matching		
CPID Matching		
Calling Number: *****	Called Number: *****	Redirect Number: *
Calling Name: *****	Called Name: *****	Redirect Name: *

Outbound Routes		
Device Selection		
Outbound Destination: **VoIP**	Host Group: **All to 172.13.4.3**	
CPID Manipulation		
Calling Number: **S**	Called Number: **D**	Redirect Number: **R**
Calling Name: **S**	Called Name: **D**	Redirect Name: **R**
Select Primary/Alternate Route		
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4		

5. Set up Inbound VoIP Rules.

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**All VoIP Requests**	**Any**	***** (1)

Inbound VoIP Request Matching					
CPID Matching					
Calling Number	*****	Called Number	*****	Redirect Number	*
Calling Name	*****	Called Name	*****	Redirect Name	*

Outbound Routes				
Device Selection				
Outbound Destination	**TDM**	Trunk Group	**Any TDM channel**	
CPID Manipulation				
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route				
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4				

Note: An asterisk specifies all VoIP addresses.

6. Validate Routing Table.

The Routing Table can be validated using the **Tests -> Router** web page.

- Navigate to **Tests -> Router** and enter data as per the following table.

Inbound Route	
• Inbound TDM	○ Inbound VoIP
Request Type	**Call**
Port	**1**
Channel	**1**
Calling Number	**156**
Calling Name	**Bill**
Called Number	**213**
Called Name	**Mary**
Redirect Number	
Redirect Name	

- Click on "Simulate Route". The **Test Results** table (bottom of page) should be filled in as follows:

Test Results	
Results	**Test Passed**
Reason	
Inbound Rule	**All Inbound TDM**
Outbound Group	**All To 172.13.4.3**

- The **Outbound Route** table should be filled in as follows:

Outbound Route	
VoIP	Device
Bridged	Method
172.13.4.3	Host
0	IP Port
156	Calling Number
Bill	Calling Name
213	Called Number
Mary	Called Name
	Redirect Number
	Redirect Name

- Try several more **Inbound Routes** varying the **Request Type, Port** and **Channel**. Since all Inbound TDM are routed using the same rule, the **Outbound Route** should be the same.
- Now test the Inbound VoIP.

Input:

Inbound Route	
○ Inbound TDM	• Inbound VoIP
Request Type	**Call**
Host	**172.13.4.3**
IP Port	
Calling Number	**100**
Calling Name	**William**
Called Number	**210**
Called Name	**Jane**
Redirect Number	**165**
Redirect Name	**Mike**

Output:

Test Results	
Results	**Test Passed**
Reason	
Inbound Rule	**All VoIP Requests**
Outbound Group	**Any TDM Channel**

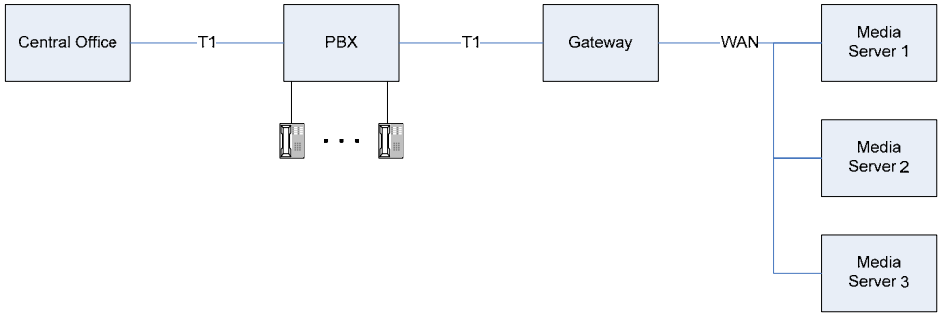
Outbound Route	
VoIP	Device
Bridged	Method
1 (see Note)	Port
1 (see Note)	Channel
100	Calling Number
William	Calling Name
210	Called Number
Jane	Called Name
165	Redirect Number
Mike	Redirect Name

Note: Since the TDM Trunk Group, **Any TDM Channel**, is **ascending – cyclic** each time you click on **Simulate Route** the port & channel is updated to the next one (with wrapping). Also, the simulation remembers where it was so if you come back to the simulation the port & channel pick up where they were. Therefore, you may not see **port = 1** and **channel = 1**, but make sure each time you click **Simulate Route** the port & channel update.

- Click on **Simulate Route** a few times **–** the port & channel should cycle through all possibilities.
- Try a few different values for **Request Type** and **Host**.
- Validation Finished.

Example #2 – Basic Call with Load Balancing

System Configuration:



Requirements:

1. All inbound TDM calls must be routed to one of the three Media Servers using Load Balancing. Media Server 1 is 172.16.3.1, Media Server 2 is 172.16.3.2 and Media Server 3 is 172.16.3.3.
2. All inbound VoIP calls must be routed out the TDM. Any available TDM channel can be used.

Steps:

1. Determine the required TDM Trunk Groups and the VoIP Host Groups.
   - From requirement #1 we have a TDM trunk group of **all TDM requests** and a VoIP host group with three hosts **172.16.3.1, 172.16.3.2** and **172.16.3.3**.
   - From requirement #2 we have a TDM trunk group of **any TDM channel.**
2. Set up TDM Trunk Groups.

Name	Selection Direction	Selection Mode	Port/Channel Content
all TDM requests			*
any TDM channel	Descending (1)	Linear (1)	*

Note: No reason for Linear/Descending just wanted to show something different.

3. Set up VoIP Host Groups.

Name	Load Balanced	Fault Tolerant	Host Summary
Load Balanced	true	false	172.16.3.1
			172.16.3.2
			172.16.3.3

4. Set up Inbound TDM Rules.

Inbound TDM Rules			
Enable	Rule Label	Request Type	Trunk Group
X	**All Inbound TDM**	**Any**	**All TDM Requests**

Inbound TDM Request Matching					
CPID Matching					
Calling Number	*	Called Number	*	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes					
Device Selection					
Outbound Destination	**VoIP**	Host Group	**Load Balanced**		
CPID Manipulation					
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route					

- **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4

5. Set up Inbound VoIP Rules.

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**All Inbound VoIP**	**Any**	*****

Inbound VoIP Request Matching					
CPID Matching					
Calling Number	*****	Called Number	*****	Redirect Number	*
Calling Name	*****	Called Name	*****	Redirect Name	*

Outbound Routes				
Device Selection				
Outbound Destination	**TDM**	Trunk Group	**Any TDM channel**	
CPID Manipulation				
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route				
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4				

6. Validate the Routing Table.

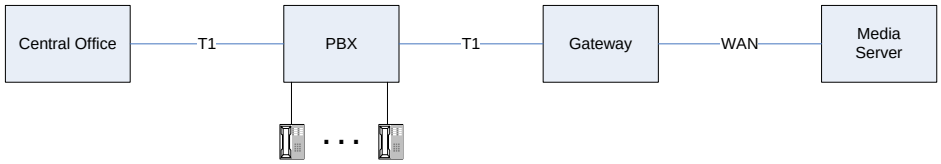
   The only difference from Example #1 is when simulating Inbound TDM calls.

   Since Example #2 uses load balancing, each time you click on **Simulate Route** the Outbound Route's Host should change. The following table shows this.

Outbound Route	
VoIP	Device
Bridged	Method
cycles through these **172.16.3.1** **172.16.3.2** **172.16.3.3**	Host
0	IP Port
156	Calling Number
Bill	Calling Name
213	Called Number
Mary	Called Name
	Redirect Number
	Redirect Name

Example #3 – Basic Call with CPID manipulation

System Configuration:



Requirements:

Routing Requirements:

1. All inbound TDM calls must be routed to the media server (IP address 172.13.4.3).
2. All inbound VoIP calls must be routed out the TDM. Any available TDM channel can be used.

CPID manipulation Requirements:

Inbound VoIP calls:

- The VoIP calls use E.164 format. We must translate from E.164 to format acceptable by a T1 line.
- Calling & Called Numbers of the form +1YYYxxxxxxx (where YYY is the area code and xxxxxxx is the phone number) must be changed to 91YYYxxxxxxx.
- Calling & Called Number of the form +ZZxxxxxxxx (where ZZ is the country code and xxxxxxxx is the phone number) must be changed to 9011ZZxxxxxxxx.

Inbound TDM calls:

- We must translate from the T1 format to E.164.
- Calling & Called Number of the form YYYxxxxxxx must be changed to +YYYxxxxxxx.
- Calling & Called Number of the form ZZxxxxxxxx must be changed to +ZZxxxxxxxx.

Steps:

1. Determine the required TDM Trunk Groups and the VoIP Host Groups.
   - From requirement #1 we have a TDM trunk group of “all TDM requests” and a VoIP host group of “172.13.4.3”.
   - From requirement #2 we have a TDM trunk group of “any TDM channel”.
2. Set up TDM Trunk Groups.

Name	Selection Direction	Selection Mode	Port/Channel Content
all TDM requests			*
any TDM channel	Ascending	Cyclic	*

3. Set up VoIP Host Groups.

Name	Load Balanced	Fault Tolerant	Host Summary
All to 172.13.4.3	false	false	172.13.4.3

4. Set up Inbound TDM Rules.

There is a rule for every CPID manipulation.

Inbound TDM Rule #1:

CPID Manipulation: **CPID of the form YYYxxxxxxx must be changed to +YYYxxxxxxx** and **CPID of the form ZZxxxxxxxx must be changed to ZZxxxxxxxx** both map to the same rule. That is, **prefix the number with a plus sign**.

Inbound TDM Rules			
Enable	Rule Label	Request Type	Trunk Group
X	**Country Code One**	**Any**	**All TDM Requests**

Inbound TDM Request Matching					
CPID Matching					
Calling Number	*	Called Number	*	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes				
Device Selection				
Outbound Destination	**VoIP**	Host Group	**All To 172.13.4.3**	
CPID Manipulation				
Calling Number	**“+” + S**	Called Number	**“+” + D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route				
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4				

5. Set up Inbound VoIP Rules.

There is a rule for every CPID manipulation plus an optional catch all.

Inbound VoIP Rule #1:

CPID Manipulation: **CPID of the form +1YYYxxxxxxx must be changed to 91YYYxxxxxxx**.

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**Country Code One** (1)	**Any**	*****

Inbound VoIP Request Matching		
CPID Matching		
Calling Number **+1.**	Called Number **+1.**	Redirect Number *****
Calling Name *****	Called Name *****	Redirect Name *****

Outbound Routes		
Device Selection		
Outbound Destination **TDM**	Trunk Group **Any TDM channel**	
CPID Manipulation		
Calling Number **"9" + lrem(S,1)**	Called Number **"9" + lrem(D,1)**	Redirect Number **R**
Calling Name **S**	Called Name **D**	Redirect Name **R**
Select Primary/Alternate Route		
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4		

Notes: A label for an Inbound VoIP rule can be the same as a label for an Inbound TDM rule.

Inbound VoIP Rule #2:

CPID Manipulation: **CPID of the form +ZZxxxxxxxx must be changed to 9011ZZxxxxxxxx**.

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**Country Code One** (1)	**Any**	*
X	**Other Country Codes**	**Any**	*****

Inbound VoIP Request Matching					
CPID Matching					
Calling Number	*****	Called Number	**+.**	Redirect Number	*
Calling Name	*****	Called Name	*****	Redirect Name	*

Outbound Routes					
Device Selection					
Outbound Destination	**TDM**	Trunk Group	**Any TDM channel**		
CPID Manipulation					
Calling Number	**"9011" + lrem(S,1)**	Called Number	**"9011" + lrem(D,1)**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route					

- **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4

Inbound VoIP Rule #3:
An optional Catch All rule.

This rule does not correspond to any required CPID manipulation. It is a catch all rule that is invoked if all the previous rules fail. This rule is optional but if you do not use a catch all rule, any inbound VoIP calls that do not match a rule are dropped.

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**Country Code One** (1)	**Any**	*
X	**Other Country Codes**	**Any**	*
X	**Catch All**	**Any**	*****

Inbound VoIP Request Matching					
CPID Matching					
Calling Number	*	Called Number	* (1)	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes				
Device Selection				
Outbound Destination	**TDM**	Trunk Group	**Any TDM channel**	
CPID Manipulation				
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route				
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4				

Notes: Since this rule is last, it matches any called number that does not start with a plus sign.

6. Validate the Routing Table.
   - For **Inbound TDM** CPID of the form 1YYYxxxxxxx is changed to +1YYYxxxxxxx.

     Input:

Inbound Route	
• Inbound TDM	○ Inbound VoIP
Request Type	**Call**
Port	**1**
Channel	**1**
Calling Number	**5553000**
Calling Name	
Called Number	**9145552345**
Called Name	
Redirect Number	
Redirect Name	

     Output:

Test Results	
Results	**Test Passed**
Reason	
Inbound Rule	**Country Code One**
Outbound Group	**All TDM Requests**

Outbound Route	
VoIP	Device
Bridged	Method
172.13.4.3	Host
0	IP Port
+5553000	Calling Number
	Calling Name
+9145552345	Called Number
	Called Name
	Redirect Number
	Redirect Name

- For **Inbound VoIP** CPID of the form +1YYYxxxxxxx is changed to 91YYYxxxxxxx.

  Input:

Inbound Route	
○ Inbound TDM	• Inbound VoIP
Request Type	**Call**
Host	**172.16.5.4**
IP Port	
Calling Number	**100**
Calling Name	**William**
Called Number	**+17165551000**
Called Name	**Jane**
Redirect Number	**165**
Redirect Name	**Mike**

  Output:

Test Results	
Results	**Test Passed**
Reason	
Inbound Rule	**Country Code One**
Outbound Group	**Any TDM Channel**

Outbound Route	
VoIP	Device
Bridged	Method
1 (see Note)	Port
1 (see Note)	Channel
100	Calling Number
William	Calling Name
917165551000	Called Number
Jane	Called Name
165	Redirect Number
Mike	Redirect Name

- For **Inbound VoIP** CPID of the form +1ZZxxxxxxx is changed to 9011YYYxxxxxxx.

  Input:

  Change Called Number to **+1435559000**.

  Output:

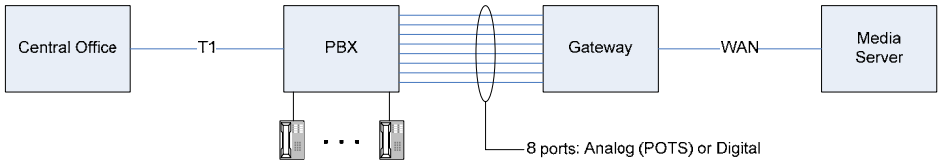
  Inbound Rule becomes **Other Country Codes**.

  Called Number becomes **+9011435559000**.

- You should validate the **Catch All** rules for Inbound TDM and VoIP by using some Called Numbers that don't match the **Country Code One** or **Other Country Code** rules.

Example #4 – Basic Call with MWI

System Configuration:



Requirements:

1. All inbound TDM requests must be routed to the media server (IP address 10.1.1.1).
2. All inbound VoIP calls must be routed out the TDM. Only TDM ports 1-6 can be used for calls.
3. All inbound VoIP MWIs (Message Waiting Indications) must be routed out the TDM. Only TDM ports 7 and 8 can be used for MWIs.

Steps:

1. Determine the required TDM Trunk Groups and the VoIP Host Groups.
   - From requirement #1 we have a TDM trunk group of "all TDM requests" and a VoIP host group of "10.1.1.1".
   - From requirement #2 we have a TDM trunk group of "TDM ports 1 to 6".
   - From requirement #3 we have a TDM trunk group of "TDM ports 7 & 8".
2. Set up TDM Trunk Groups.

Name	Selection Direction	Selection Mode	Port/Channel Content
all TDM requests			*
TDM ports 1 to 6	Ascending	Cyclic	1-6
TDM ports 7 & 8	Ascending	Cyclic	7,8

3. Set up VoIP Host Groups.

Name	Load Balanced	Fault Tolerant	Host Summary
All to 10.1.1.1	false	false	10.1.1.1

4. Set up Inbound TDM Rules.

Inbound TDM Rule #1:
For Inbound TDM Requests (requirement #1).

Inbound TDM Rules			
Enable	Rule Label	Request Type	Trunk Group
X	**All TDM Requests**	**Any**	**All TDM Requests**

Inbound TDM Request Matching					
CPID Matching					
Calling Number	*	Called Number	*	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes				
Device Selection				
Outbound Destination	**VoIP**	Host Group	**All to 10.1.1.1**	
CPID Manipulation				
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route				

- **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4

5. Set up Inbound VoIP Rules.

Inbound VoIP Rule #1:
For inbound VoIP calls (requirement #2).

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**All VoIP Calls**	**Call**	*****

Inbound VoIP Request Matching		
CPID Matching		
Calling Number *****	Called Number *****	Redirect Number *
Calling Name *****	Called Name *****	Redirect Name *

Outbound Routes		
Device Selection		
Outbound Destination **TDM**	Trunk Group **TDM ports 1 to 6**	
CPID Manipulation		
Calling Number **S**	Called Number **D**	Redirect Number **R**
Calling Name **S**	Called Name **D**	Redirect Name **R**
Select Primary/Alternate Route		
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4		

Inbound VoIP Rule #2:
For inbound VoIP MWI requests (requirement #3).

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**All VoIP Calls**	**Call**	*
X	**All VoIP MWIs**	**Message**	****

Inbound VoIP Request Matching					
CPID Matching					
Calling Number	*	Called Number	*	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes				
Device Selection				
Outbound Destination	**TDM**	Trunk Group	**TDM ports 7, 8**	
CPID Manipulation				
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route				

- **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4

6. Validate the Routing Table.
   - An exercise left to the reader.

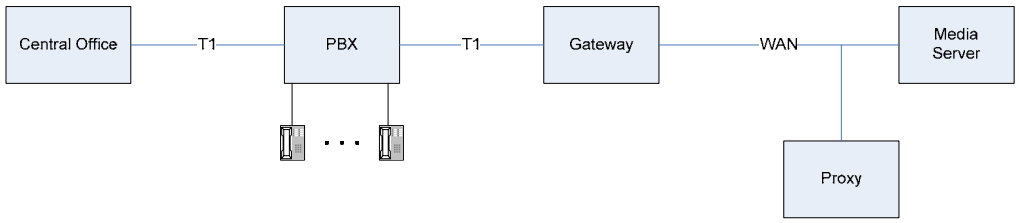
Example #5 – Basic Calls with Proxy

This example is the same as Example #1 except a Proxy is used.

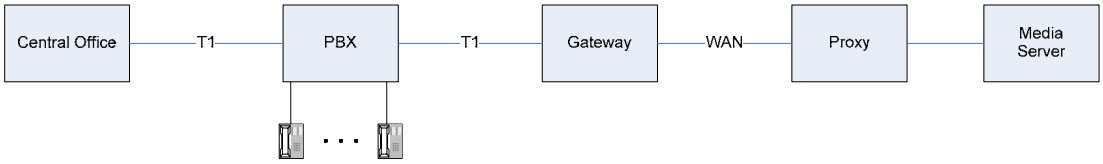
Physical System Configuration verses Logical System Configuration:

The proxy is connected to the same network as the Media Server. However, logically the proxy sits between the Gateway and the Media Server. That is, all messages from the Gateway go to the proxy and the proxy retransmits the messages to the Media Server. All messages from the Media Server go to the proxy and the proxy retransmits them to the Gateway.

Physical System Configuration:



Logical System Configuration:



Requirements:

The requirements are the same as Example #1 with the addition of using a Proxy at 172.16.100.1 port 5060.

Steps:

The steps are the same as Example #1 with the addition of setting up the proxy. The proxy setup is shown following:

Setup of Proxy:

- The Proxy is setup on the **VoIP -> General** web page, under the Proxy section.

Proxy	
Primary Proxy Server Address	172.16.100.1 (1)
Primary Proxy Server Port	5060
Backup Proxy Server Address	(2)
Backup Proxy Server Address	
Proxy Query Interval (s)	(3)

Notes:

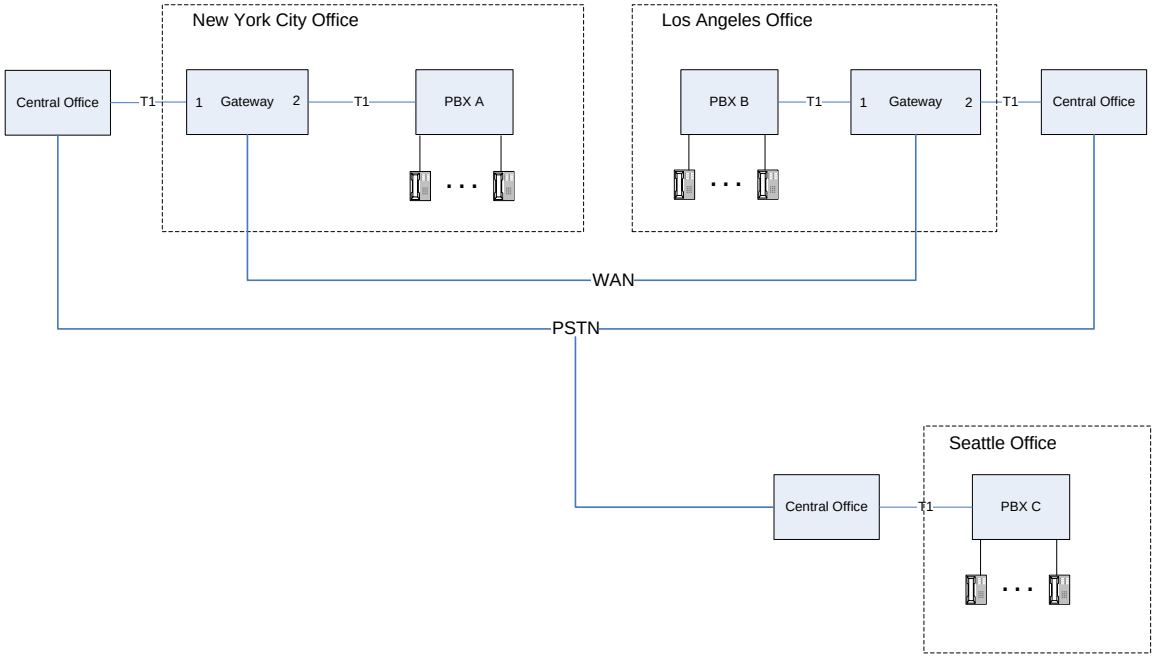
1. Leaving this entry blank denotes no proxy (as in Example #1).
2. A backup proxy can be specified. The backup is used in the event the primary proxy goes down.
3. The Proxy Query Interval is used only when a backup proxy is specified. The interval specifies the time between queries to the primary proxy. If the primary proxy does not answer the query, it is marked as down and the backup proxy is used.

Example #6 – Toll Bypass

Toll Bypass allows a company to make interoffice calls using the company's WAN and without using the public telephone network – reducing the cost of phone calls.

System Configuration:

The sample system configuration shows three offices. We named them New York City (NYC), Los Angeles (LA) and Seattle to make the example more real-world. Each office has a number of phones connected to an internal PBX. The Seattle PBX connects directly to the Central Office and on to the PSTN (public switched telephone network). The New York City and Los Angeles offices have their PBX connected to a gateway. Each gateway connects to the company's WAN and to a Central Office.



System Requirements:

- Calls between the New York City office and the Los Angeles office use the WAN.
- Calls between the New York City office and the Seattle office use the PSTN.
- Calls between the Los Angeles office and the Seattle office use the PSTN.
- In New York and Los Angeles all inbound calls from the CO (PSTN) are routed to the local PBX.
- In New York and Los Angeles all inbound calls from the WAN are routed to the local PBX. These could be calls form other offices or other companies who all use the WAN.

System Setup:

- New York City phones have extensions 1xxx.
- Los Angeles phones have extensions 2xxx.
- Seattle phones have extensions 3xxx.
- New York City gateway is at 172.16.5.4.
- Los Angeles gateway is at 172.16.5.3.

Routing Requirements:

New York City Gateway:

1. All inbound VoIP calls must be routed out the TDM span connected to the PBX.
2. All inbound TDM calls from the span connected to the CO must be routed out the TDM span connected to the PBX.
3. All inbound TDM calls from the span connected to the PBX and with a destination of 2xxx must be routed out the VoIP to the Los Angeles gateway (172.16.5.3).
4. All other inbound TDM calls from the span connected to the PBX must be routed out the TDM span connected to the CO.

Los Angeles Gateway:

1. All inbound VoIP calls must be routed out the TDM span connected to the PBX.
2. All inbound TDM calls from the span connected to the CO must be routed out the TDM span connected to the PBX.
3. All inbound TDM calls from the span connected to the PBX and with a destination of 1xxx must be routed out the VoIP to the New York City gateway (172.16.5.4).
4. All other inbound TDM calls from the span connected to the PBX must be routed out the TDM span connected to the CO.

Steps:

The steps to setup the NYC gateway are presented. From this the reader can derive the setup of the LA gateway.

1. Determine the required TDM Trunk Groups and the VoIP Host Groups.
   - From requirement #1 we have a VoIP host group of “all inbound calls” and a TDM trunk group of “span connected to PBX”.
   - From requirement #2 we have a TDM trunk group of “span connected to CO” and a TDM trunk group of “span connected to PBX”.
   - From requirement #3 we have a VoIP host group of “Los Angeles gateway”
   - From requirement #4 we have no new trunk groups or VoIP host groups.
2. Set up TDM Trunk Groups.

Name	Selection Direction	Selection Mode	Port/Channel Content
Span to PBX	Ascending	Cyclic	2
Span to CO	Ascending	Cyclic	1

3. Set up VoIP Host Groups.

Name	Load Balanced	Fault Tolerant	Host Summary
Los Angeles Gateway	false	false	172.16.5.3

4. Set up Inbound TDM Rules.

Inbound TDM Rule #1:
All inbound TDM calls from the span connected to the CO must be routed out the TDM span connected to the PBX (routing requirement #2).

Inbound TDM Rules			
Enable	Rule Label	Request Type	Trunk Group
X	**Calls From CO**	**Call**	**Span to CO**

Inbound TDM Request Matching		
CPID Matching		
Calling Number: *****	Called Number: *****	Redirect Number: *
Calling Name: *****	Called Name: *****	Redirect Name: *

Outbound Routes		
Device Selection		
Outbound Destination: **TDM**	Trunk Group: **Span to PBX**	Route Method: **Bridged**
CPID Manipulation		
Calling Number: **S**	Called Number: **D**	Redirect Number: **R**
Calling Name: **S**	Called Name: **D**	Redirect Name: **R**
Select Primary/Alternate Route		
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4		

Inbound TDM Rule #2:
All inbound TDM calls from the span connected to the PBX and with a destination of 2xxx must be routed out the VoIP to the Los Angeles gateway (172.16.5.3) (requirement #3).

Inbound TDM Rules			
Enable	Rule Label	Request Type	Trunk Group
X	**Calls From CO**	**Call**	**Span to CO**
X	**Calls From PBX To LA**	**Call**	**Span to PBX**

Inbound TDM Request Matching					
CPID Matching					
Calling Number	*****	Called Number	**2.**	Redirect Number	*
Calling Name	*****	Called Name	*****	Redirect Name	*

Outbound Routes					
Device Selection					
Outbound Destination	**VoIP**	Trunk Group	**Los Angeles Gateway**	Route Method	**Bridged**
CPID Manipulation					
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route					
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4					

Inbound TDM Rule #3:
All other inbound TDM calls from the span connected to the PBX must be routed out the TDM span connected to the CO (requirement #4).

Inbound TDM Rules			
Enable	Rule Label	Request Type	Trunk Group
X	**Calls From CO**	**Call**	**Span to CO**
X	**Calls From PBX To LA**	**Call**	**Span to PBX**
X	**Calls From PBX to PSTN**	**Call**	**Span to PBX**

Inbound TDM Request Matching					
CPID Matching					
Calling Number	*	Called Number	* (1)	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes					
Device Selection					
Outbound Destination	**TDM**	Trunk Group	**Span to CO**	Route Method	**Bridged**
CPID Manipulation					
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route					

- **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4

Notes: Recall that the rules are matched top down. So this rule is only matched when the previous rule fails. Therefore, this rule catches all calls except those to 2xxx.

5. Set up Inbound VoIP Rules.

Inbound VoIP Rule #1:
All inbound VoIP calls must be routed out the TDM span connected to the PBX (requirement #1).

Inbound VoIP Rules			
Enable	Rule Label	Request Type	Originating VoIP Host Address
X	**Inbound VoIP Calls**	**Call**	*****

Inbound VoIP Request Matching					
CPID Matching					
Calling Number	*	Called Number	*	Redirect Number	*
Calling Name	*	Called Name	*	Redirect Name	*

Outbound Routes					
Device Selection					
Outbound Destination	**TDM**	Trunk Group	**Span to PBX**		
CPID Manipulation					
Calling Number	**S**	Called Number	**D**	Redirect Number	**R**
Calling Name	**S**	Called Name	**D**	Redirect Name	**R**
Select Primary/Alternate Route					
• **Primary** ○ Alt-1 ○ Alt-2 ○ Alt-3 ○ Alt-4					

Media Gateway Parsers 6

This section describes the Dialogic® Media Gateway in-band Type I (on-hook) and Type II (off-hook) integration parsers for analog and T1 CAS integrations, and the display parsers for digital integrations. These parsers allow the user to define the meaning of either the in-band/on-hook integration strings or display strings received from the telephony network. For this discussion, the term parser will be used when referencing all three variants listed above. Various options for entering the configuration data, parser syntax rules, and several examples are presented.

Note: The information in this section applies to all models of the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) models using the CAS protocol.

Information about the Media Gateway parsers is included in the following sections:

- Configuration Options 247
- Parsing Configuration Syntax 248

The parser allows the user to enter rules that define the meaning of the strings that can be received from the telephony network. The Media Gateway will use these rules to parse the strings and extract the source party information, destination party information, call reason (direct, busy, ring-no-answer, etc.), and call origin (internal or external). Note that the default rules supplied for the display parsing on digital units handle most scenarios, but the analog CPID parsing default rules are only included to provide examples for the user.

The user configurable rules (collectively known as the configuration data) contain expressions that represent the various types of integration strings available on the telephony switch interface. The configuration data contains rules and expressions that specify the location of the source, destination, reason, and origin information in the integration or display string.

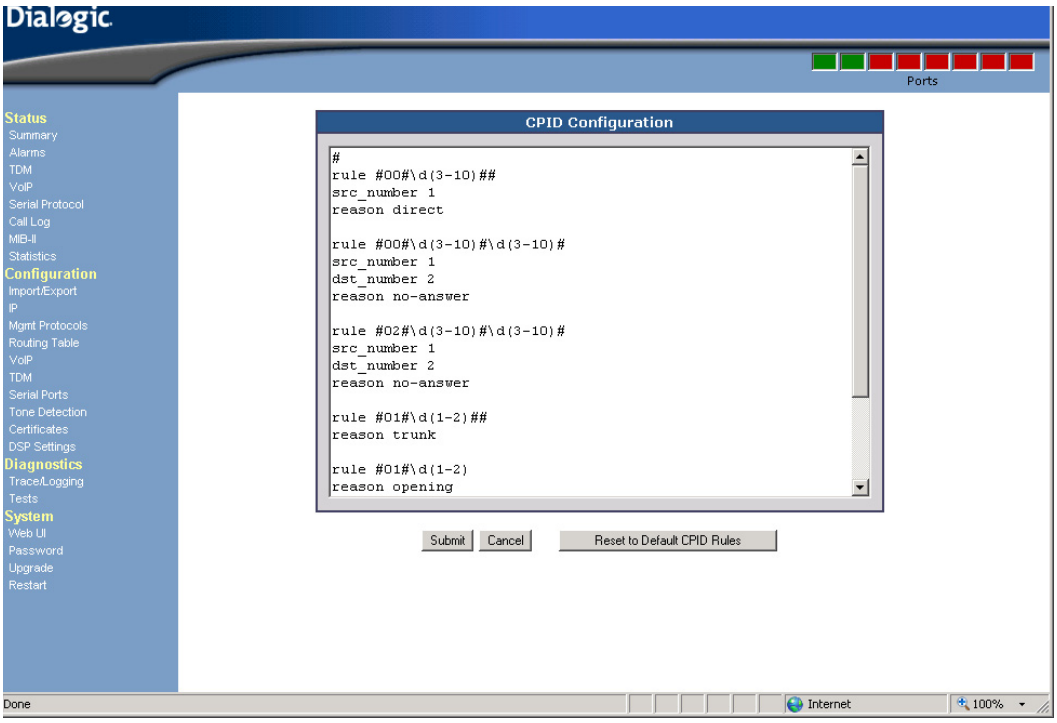
6.1 Configuration Options

The user can access the parsing configuration data in several different ways:

- Media Gateway configuration file (*.ini) which allows the user to use the import/export Web page to upload new configuration data or to download existing configuration data for review.
- Text file (must have an .adt file extension) containing only the configuration data.
- The cpid.htm hidden Web page.
- For type I and II analog parsing rules only, the analog page on the Web interface.

Figure 41 shows the default analog parsing configuration (used for either type I or II CPID) on the analog Web page. Figure 42 shows this same data as it appears in the *.ini file. Figure 43 shows a sample, downloadable text based *.adt file. Figure 44 shows the generic CPID configuration page available to all emulating units with the example showing the default Mitel digital display parsing rules.

Figure 41. Default Analog CPID Configuration Data on Analog Web Page



6.2 Parsing Configuration Syntax

The configuration data syntax is used to describe the type I or II CPID (analog or T1 CAS) or display (digital) strings that may be received from the telephone network. The type of information that can be received using analog DTMF integrations is a subset of that which can be gleaned from digital displays. For this reason, the rule syntax used for analog parsing is a much simpler subset of the digital parsing syntax.

The overall rule syntax is a subset of PERL. The syntax supports the extraction of the following information:

- Source Party Information (may include ANI information) - for analog DTMF integrations, this can only be numeric, whereas for digital integrations, the information can be a number and/or a name.
- Destination Party Information (may include DNIS information) - for analog DTMF integrations, this can only be numeric, whereas for digital integrations, the information can be a number and/or a name.

Figure 42. Default Analog CPID Configuration in the .ini File

```
;Version Information:
;MAC:00-a0-e6-06-05-07
;IP:10.0.1.41
;DSP Firmware: |9.1w/Fax|FRI MAY 02 17:59:51 2003|
;DSP Firmware (ROM): 9.1 w/Fax|FRI MAY 02 17:59:51 2003
;Main Board Boot (ROM): |4.2||WED APR 02 17:51:55 2003
;Gateway Application (ROM): |lab||FRI JUL 02 14:18:13 2004
;Main Board CPLD: Platform 3 Ver 1
;Adept Config: Default
;*********************************************************************
;Client IP Address: IP Adress in dotted decimal notation
ipClientAddr = 10.0.1.41
  *
  *
  *
;Call Progress Cadence Time Deviation (ms):Number between 0 - 1000
cpToneTimeDeviation: Tone Definition1 Time1 = 0
cpToneTimeDeviation: Tone Definition1 Time2 = 0
cpToneTimeDeviation: Tone Definition1 Time3 = 0
  *
  *
  *
cpToneTimeDeviation: Tone Definition16 Time1 = 0
cpToneTimeDeviation: Tone Definition16 Time2 = 0
cpToneTimeDeviation: Tone Definition16 Time3 = 0

;-CPID RULES
# 95551212 (direct external call)
rule 9\d(2-10)
src_number 1
reason direct

# *8** (disconnect string)
rule \*8\*\*
reason disconnect

# 3211 (forwarded internal call to 3211)
rule \d(2-6)
dst_number 1
reason no-answer
```

- Reason Code (no-answer, busy, etc.)
- Call Origin (internal, external)

The configuration data consists of call-class parsing rules. Call-class parsing rules are PERL-like expressions that define the incoming integration or text strings. The configuration data may contain multiple rules, each representing a different type of integration or display string in which the call party information exists in a different format (either textual representation or delimited fields).

Figure 43. Sample Analog Type II CPID Configuration Data in the .adt File

```
#### ATT System 25
rule #00#\d(1-5)##
src_number 1

rule #00#\d(1-5)#\d(1-5)#
src_number 1
dst_number 2
reason no-answer

rule #01#\d(1-3)##
reason trunk

rule #01###
reason opening

rule #01##
reason opening

rule #01#
reason opening

rule #02#\d(1-5)#\d(1-5)#
src_number 1
dst_number 2
reason no-answer

rule #03##\d(1-5)#
dst_number 1
reason no-answer

rule #04##\d(1-3)#
reason trunk

rule #05#\d(0-10)#\d(0-10)
reason disconnect
```

Figure 44. Default Mitel Digital CPID Configuration Data (cpid.htm)

CPID Parsing Rules

```
; MITEL


; Reason translations
tran reason |BUSY|busy
tran reason |NO ANS|no-answer
tran reason |DO NOT DISTURB|no-answer
tran reason |ALWAYS|fwd-all
tran reason |FWD|fwd-all
tran reason |FORWARD|fwd-all
tran reason |FORWARDED|fwd-all
tran reason |default|direct
tran origin |default|internal

; time in first row, ignore these
rule .*\d+:\d+.*
reason time

; TRUNK 102 IS CALLING  FWD FROM JOEY ALWAYS
rule \bTRUNK\b\d*\b\D*\bFROM\b\d*\b\w*\b~*\b
src_name Outside_Call
dst_number 2
dst_name 2
reason 1
origin external

; JOE IS CALLING        FWD FROM JOEY ALWAYS
rule \b\d*\b\w*\bIS\b\D*\bFROM\b\d*\b\w*\b~*\b
src_number 1
src_name 1
dst_number 2
dst_name 2
reason 1

; 455 IS CALLING
; JOE IS CALLING blah blah blah
; JOE IS RINGING YOU BACK
rule \b\d*\b\w*\bIS\b\D*
src_number 1
```

The parser attempts to match an input integration or display string to a rule defined by the configuration data. If a match is made, the parser uses the call-party, reason code, and origin specifiers of the matching rule. In this manner, the parser can extract the call party information from the correct locations of the integration or display strings.

6.2.1 Display Translation Descriptors

Translation descriptors define translations between telephone switch-specific display tokens and strings that the application uses. Translation descriptors are global translations that govern all display parsing rules. The following is a reason code example.

```
trans reason |FWD|fwd-all
```

The above provides a translation of the switch-specific reason code “FWD” to the application string “fwd-all”. If the token “FWD” is found in the reason-code section of a display, the reason code presented to the application will be “fwd-all”. This provides a means of normalizing different switch-specific reason codes into strings that can be recognized by the application. The following is an origin example.

```
trans origin |default|internal
```

If the switch-specific code is ‘default’, then the application string of the translation is set as the default translation result if no switch-specific string is found. In the above case, the default call origin for all displays is 'internal'.

6.2.2 Call Class Rules

On a telephone network, there can be several different string formats for call information. Integration or display strings which use the same format are said to be in the same “Call Class”. That is, two integration or display strings can be parsed using the same rules if they are in the same call class. If an integration or display string cannot be parsed using the same rules, then a new call class must be declared.

Regular Expressions

Call classes will be recognized using user-defined regular expressions. The parser uses standard regular expression “metacharacters” (special characters which are used to describe sequences of regular characters) in addition to some specific metacharacters.

Table 16. Parser Regular Expressions

Characters and Meta-Characters	Purpose	Analog Parsing Applicable
.	Any single character.	Yes
\	Character to right is literal. (quoted character)	Yes
?	0 or 1 of preceding character.	No
*	0 or more of preceding character.	Yes
+	1 or more of preceding character.	No
~	A reason code or call origin character.	No
\d	Digit character. (call party number)	Yes
\D	Non-digit character.	No
\w	Word character. (call party name) Must start with alpha, then can be alpha, digit, -, comma, space, ., or _.	No
\s	Whitespace character. Space, newline, or tab.	No
\b	Word boundary. Whitespace, punctuation, beginning or end of text. Used to specify that strings are bounded at start or end of string. Not explicitly searched for. Checked on reasons, digits, and literal strings.	No

Table 16. Parser Regular Expressions (Continued)

Characters and Meta-Characters	Purpose	Analog Parsing Applicable
\nnn	Octal control character.	No
(min,max)	Braces. Follows other chars. If non-null, length must be within min/max.	Yes

Rule Syntax

A call-class rule starts with the tag “rule”. All characters following the tag define the rule. Following the rule are specifiers that define the location in the integration or display string of the call party information.

For example, to describe the integration string:

```
[A1500*300#]
```

the rule might read:

```
rule A1\d(0-10)\*\d(0-10)*#
dst_number      1
src_number      2
reason          no-answer
```

This means that when the regular expression is satisfied, the first digits string is the destination number, the second digit string is the source number, and the reason code is 'no-answer'.

In order to describe the display:

```
[a= JOE 123 to BILL 456 b]
```

the rule might read:

```
rule .= \w*\s\d* to \w*\s\d*\s*~*
src_number      1
src_name        1
dst_number      2
dst_name        2
reason          1
origin          internal
```

This means that when the regular expression is satisfied, the first digits string is the source number, the second digits string is the destination number, the first word string is the source name, the second word string is the destination name, and the reason code is at the end. The origin of the call is “internal”.

Rule Order

The parser attempts to match an input integration or display string with a rule contained in the configuration data. The rules are compared to the specified string from the top rule to the bottom rule. Because of this, rules that contain the most specific information should be listed in the configuration file first. In this way, exact matches can occur before a more generic 'catch-all' rule is reached.

If a rule does not match, the parser attempts to match the next rule in the list. If no rules match, then the integration parsing fails. For this reason, the last rule in the rule list should be a very generic rule that provides the best possibility of extracting the desired information from the integration string.

Reason Tokens

The reason token specified in a rule is translated by the DMG1000 into corresponding IP call information. The following reason token strings specified in a rule are recognized by the DMG1000:

Table 17. Parser Reason Codes

Reason String	Used by Media Gateway
no-answer	Call is tagged as a forwarded on no-answer call.
busy	Call is tagged as a forwarded on busy call.
direct	Call is tagged as a direct call.
fwd-all	Call is tagged as a forwarded all call.
disconnect	Call is disconnected.

Any other call reason token specified in a rule will cause the Media Gateway to ignore the received string (if the received string matches the rule).

Data Security 7

Information about data security and how it is supported by the Dialogic® Media Gateway is described in the following sections:

- Data Security Overview 255
- Secure HTTP 255
- SIP Call Control Security using TLS 257
- Secure Voice Data 260
- Installing Certificate Using Internet Explorer 262

7.1 Data Security Overview

Data security on the Media Gateway includes the use of various secure protocols when transmitting and receiving data. The Media Gateway secures three types of data:

- HTTP - The data transmitted between the Media Gateway and a Web browser. To secure HTTP, the Media Gateway uses the Secure HTTP (HTTPS) protocol.
- Call Control - The data used to setup and tear down a call. To secure Call Control, the Media Gateway uses Transport Layer Security (TLS) on top of SIP
- Voice - The actual conversation once a call is connected. To secure voice, the Media Gateway uses Secure RTP (SRTP).

The HTTPS and TLS protocols require digital identity certificates (e.g. public key certificates). Therefore, certificate management is also covered in this section.

7.2 Secure HTTP

HTTP data is transmitted as messages between the Media Gateway and a Web browser. These messages travel on the network as clear text and can be “listened” to by anyone. Even though the HTTP interface has access security (via a password), privacy is not secure.

As an example, if a message containing a request to change a password were captured by a hacker or third party, the hacker or third party could log on to the Media Gateway and change the configuration. HTTPS safeguards HTTP data by encryption and authentication. With HTTPS, messages are no longer transmitted as clear text and are not readily readable.

HTTPS requires two actions by the user:

- Both the Media Gateway and the PC on which the Web browser used to connect to the Media Gateway via HTTPS is running must be configured with the proper certificate.

- When accessing the Media Gateway, use https:// instead of the non-secure http:// followed by the Media Gateway's URL.

This section includes the following information about HTTP security:

- HTTPS Certificate Configuration
- HTTPS Example

7.2.1 HTTPS Certificate Configuration

An HTTPS certificate can be either self-signed or certificate authority (CA) signed. A self-signed certificate can be generated by the Media Gateway. CA signed certificates must be requested by the Media Gateway and then signed by a CA.

When using a self-signed certificate:

- The Media Gateway generates a self-signed public key certificate.
- This certificate is then exported and downloaded from the Media Gateway to a PC via HTTP (or HTTPS if already active).
- The certificate is then configured into the Windows® PC running the HTTPS Web browser used to connect to the Media Gateway.
- From this PC, the user logs on to the Media Gateway using the https://[URL].
- HTTPS is then automatically used when accessing all subsequent Web pages.

When using a CA signed certificate:

- The Media Gateway generates a certificate signature request (CSR).
- The CSR is exported from the Media Gateway to a PC via HTTP (or HTTPS if already active).
- The CSR is used by the CA to create a signed certificate.
- The CA signed certificate is uploaded to the Media Gateway.
- The root certificate of the CA that signed the CSR is configured into the PC running the Web browser used to connect to the Media Gateway via HTTPS.
- The user logs into the Media Gateway by going to https://[URL]
- HTTPS is automatically used when accessing all the subsequent Web pages

The choice of either self-signed or CA-signed certificates depends on the system administration and the desired level of trust within the system. Self-signed certificates are generated by the Media Gateway and therefore do not cost any money - and may require less time to install. A self-signed certificate is simply downloaded from the gateway and installed on the PC running the Web browser used to connect to the Media Gateway via HTTPS.

However, when self-signed certificates are used, the PC/Web Browser must have a unique certificate installed for each Media Gateway with which it will communicate. This process could get lengthy if the PC/Web Browser needs to communicate with a number of Media Gateway units. On the other hand, CA signed certificates require time and effort to install since the certificates must be signed by a CA. However, once you have the signed certificate, the CA root certificate can be used to communicate with multiple Media Gateway units.

7.2.2 HTTPS Example

An example of how HTTPS is used with a self-signed certificate is described below. In the example, the Media Gateway has an IP address of 172.16.3.10 and uses a self-signed certificate.

1. Start the Media Gateway.
2. Start Internet Explorer (or any Web browser that supports HTTPS).
3. In the Web browser Address box, enter http://172.16.3.10.
4. At the login screen, enter a User name and Password, and click OK to login to the Media Gateway.
5. Select the Security Web page > Certificate Management tab > Certificate Usage table > HTTPS parameter and check that **Self Signed** is the selected value.
6. Go to the HTTPS table and click on Generate button on the Self Signed row. The Self Signed Certificate Generation screen will appear.
7. Fill in the text boxes in the Value row of both the Certificate X509 Extensions and Certificate Subject tables and then click the Generate button. After a short time, the following message will appear: “Self signed Certificate was created”. Click Continue to return to the Certificate Management Web page.
8. In the HTTPS table, click the Export button in the **Action** column of the **Self Signed** row to download the certificate from the Media Gateway to the PC.
9. Configure this certificate on the PC running the Web browser used to connect to the Media Gateway via HTTPS. See Section 7.5, “Installing Certificate Using Internet Explorer”, on page 262 for details.
10. In the Web browser Address box, enter https://172.16.3.10.
11. At the login screen, enter a User name and Password. Then click OK to login to the Media Gateway. HTTPS is now active.

7.3 SIP Call Control Security using TLS

This section includes the following information about SIP Call Control security using TLS:

- TLS Certificate Configuration
- TLS Feature Configuration
- TLS Examples

SIP is an application protocol used for VoIP call control. SIP messages are used for call setup and tear down. These messages contain information such as call-party information, call media type, whether it is a secure call, and if so, what encryption algorithm is used, etc. The SIP protocol can be carried by UDP, TCP, or TLS transports. Both UDP and TCP transport data in clear text. As a result, UDP and TCP can easily be monitored by third party hackers. TLS, on the other hand, carries SIP data in a secure way by encrypting the data and authenticating the transport connections. Authentication guarantees that you are talking to the intended peer.

7.3.1 TLS Certificate Configuration

A TLS certificate can be self-signed or certificate authority (CA) signed. A self-signed certificate can be generated by the Media Gateway. CA signed certificates must be requested by the Media Gateway and signed by a CA.

When using a self-signed certificate:

- The Media Gateway generates a certificate which will be installed on VoIP devices that will communicate with the Media Gateway via TLS.

When using a CA signed certificate:

- The Media Gateway generates a certificate signature request (CSR) to a PC.
- The CSR is used by the CA to create a signed certificate.
- The root certificate of the CA that signed the CSR is uploaded to the Media Gateway along with the CA signed certificate.
- The root certificate of the CA that signed the CSR, as well as the signed certificate, are also configured into the VoIP devices that will communicate with the Media Gateway via TLS.

The choice of either self-signed or CA-signed certificates depends on the system-administration and the desired level of trust within the system. Self-signed certificates are generated by the Media Gateway and therefore, do not cost any money - and may take less time to install. A self-signed certificate is simply downloaded from the gateway and installed on VoIP devices that will communicate with the Media Gateway via TLS. However, when self-signed certificates are used the VoIP device must have a unique certificate installed for each Media Gateway with which it will communicate. This process could become lengthy if the VoIP device needs to communicate with a number of Media Gateway units. On the other hand, CA signed certificates require time and effort since the certificates must be signed by a CA. However, once you have the signed certificate, the CA root certificate can be used to communicate with multiple Media Gateway units.

7.3.2 TLS Feature Configuration

TLS has the following configurable features. The values of these configuration parameters can usually be left as default.

- SNTP Server IP Address - A server that the Media Gateway gets current time from to compare to the expiration date of a certificate. This is how the Media Gateway identifies an expired certificate when necessary. The expired certificates are identified by certificate date verification. This time-providing server is needed if a TLS certificate date is verified.
- TLS Transport Enabled - This parameter enables use of the TLS protocol and must be set to 'Yes'.
- TLS Server Port - This is the IP port post number to listen to for TLS connection requests. Any number between 1024 and 65000 is valid. The default is 5061. If you wish to use a port number other than the default, specify the number. The Media Gateway will then communicate this number to peers via URI.

- TLS Cipher List - The Cipher list is not a configurable parameter. The Media Gateway supports 6 ciphers in a list but cannot be changed. Valid OpenSSL ciphers can be found at: *http://www.openssl.org/docs/apps/ciphers.html*

  A default cipher list must be specified for TLS to work. The Media Gateway uses the default: ALL:!ADH:!LOW:!EXP:!MD5:@STRENGTH.
- SSL TLS Protocol - This parameter specifies the SSL record type to be used with the TLS connections and can be set to use SSLv3 and/or TLSv1.
- Mutual Authentication Required - Mutual Authentication Required. This parameter should be set to 'Yes' if the user wants the Media Gateway to authenticate the VoIP endpoint that it is communicating with when the VoIP endpoint initiates a SIP session. Otherwise, this parameter can be set to 'No'.
- TLS Inactivity Timer - This will determine when to close a TLS port. Any number between 10 to 60000 milliseconds is valid.
- SIPS URI Scheme Enabled - Selects the URI scheme, SIP or SIPS, that the Media Gateway will use for outgoing SIP call-requests. This may be limited by the capability of the other party that the Media Gateway communicates with. The Media Gateway accepts both SIP and SIPS URI schemes.
- Verify TLS Peer Certificate Date - If enabled, the peer certificate date is verified to detect if the peer certificate has expired. If so, the call request will be rejected. This indicates whether or not the certificate date is verified. Enable this feature if you want to detect expired certificates. Otherwise, keep it disabled. This feature will work correctly only if a SNTP server is available.
- Verify TLS Peer Certificate Trust - This indicates whether or not a certificate trust is verified. A certificate trust is the identity that signs the certificates. If the Media Gateway only accepts certificates signed by certain CA, then the Media Gateway compares the trust on a certificate to its trust list. If the trust is found in the list then the verification will pass. Enable this feature to increase security. However, if the other party that the Media Gateway communicates with is not capable of generating a trust, this feature must be disabled to avoid verification failure. You may also leave this feature disabled if encrypting data is sufficient.

7.3.3 TLS Examples

The following examples show how the Media Gateway should be configured for using TLS.

Example 1:

Assume the Media Gateway is talking to an IP phone that requires TLS and supports SIPS URI.

The Media Gateway can be configured as follows:

- SNTP Server IP Address: Leave blank
- TLS Inactivity Timer: Use default value
- TLS Server port: Use default value
- SIPS URI Scheme Enable: Yes
- Cipher List: Use default value
- Verify TLS Peer Certificate date: No

- Verify TLS Peer Certificate trust: No

Example 2:

Assume a company has five sites: one in Indiana, one in Illinois, one in California, one in New York, and one in Washington. Each of these sites uses a local telephone company. People at different sites frequently call each other. They have decided to use VoIP service with one Media Gateway at each site. They also use one media server. A CA signed certificate, which will expire in 1 year, is used. There is no convenient SNTP server. Each of the five Media Gateways can be configured as follows:

- SNTP Server IP Address: Leave blank
- TLS Inactivity Timer: Use default value
- TLS Server port: Use default value
- SIPS URI Scheme Enabled: Yes
- Cipher List: Use default value
- Verify TLS Peer Certificate date: No
- Verify TLS Peer Certificate trust: No

7.4 Secure Voice Data

This section includes the following information about secure voice data:

- Configuration
- Secure Voice Data Examples

Once a Voice over IP (VoIP) call is established, voice data is transported in the form of RTP packets. The voice data can be easily extracted from RTP packets and replayed using commercially available software. SRTP adds security by encrypting voice data and authenticating packets.

The two parties involved in a conversation must be “compatible” in the sense that each party understands the other party's cipher requirements and supports them. Configuration provides the following benefits:

- Support for more devices - For example, Media Gateway can talk to a device that supports either SHA1 32 bits authentication tag or SHA1 80 bits authentication tag.
- Turn security on or off completely. This allows the Media Gateway to talk to a device that doesn't support security at all.

7.4.1 Configuration

Secure RTP (SRTP) includes the following configuration parameters:

- SRTP Preference - Values for this parameter are SRTP_Only, RTP_Only, or SRTP_Preferred. A single setting applies to all channels. If SRTP_Only is specified, the gateway will only request secure audio and will reject all requests for non-secure audio. If SRTP_Preferred is specified, the gateway will request both secure audio and non-secure audio, with a preference

for secure audio, and the gateway will accept requests for both secure and non-secure audio. If RTP_Only is specified, the gateway will only request non-secure audio and will only accept requests for non-secure audio.

Note: SRTP_Preferred is not supported in Version 5.1 SU2 Software.

- Master Key Index (MKI) on Transmit Stream - With SRTP, audio data for a conversation is encrypted using a key that is called the session key. Each session key is derived from a master key communicated through SIP SDP. Multiple master keys may be used to add security. When multiple master keys are used, the master key index is used to identify a master key. When an audio data packet arrives, the master key index is specified in the packet so the packet receiver knows which key to use to decrypt the packet. Not all the devices support this feature.
- Key Derivation Enable - Key derivation refers to the process used to generate a session key from a master key. If the session key is generated once for each conversation, Key Derivation Enabled is set to false. Otherwise, if a session key is generated more than once from a master key, Key Derivation Enabled is set to true. Enabling Key Derivation provides more security, but not all devices support this feature.
- Key Derivation Rate (KDR) - This parameter is only usable if Key Derivation Enabled is set to Yes. KDR takes values of 16 to 24. When 0 is specified, the session key is only derived once, which is the same as when Key derivation Enabled set to No. However, if key derivation rate is to a value greater than 0, a new session key from the same master key is derived whenever the audio data packet index reaches the multiple of 2^{KDR}. For example, if KDR is 16, $2^{16} = 65536$. The session key will be derived whenever a packet index reaches 65536, 2*65536, 3*65536, etc.
- Cipher Mode - Cipher is the algorithm used to encrypt/decrypt a packet. The Media Gateway supports plain text or AES counter mode. When plain text is specified, no encryption/decryption is performed on the audio data and Authentication is skipped. By default, the Media Gateway supports the AES counter mode.

  Note: Plain_Text is not supported in Version 5.1 SU2 Software.
- Authentication Type - The Media Gateway supports no authentication or SHA1. It is recommended that you use authentication whenever the cipher is non-null as it provides more security.
- Authentication tag length. When SHA1 is used, the tag can be 32 bits or 80 bits. By default, the Media Gateway supports SHA1 80 bits.

7.4.2 Secure Voice Data Examples

The following examples show how the Media Gateway should be configured for SRTP.

Example1:

In this example, the Media Gateway is talking to an IP phone that requires security and supports the following:

- Cipher: AES Counter Mode
- Authentication: SHA1 32 bits or 80 bits
- MKI: Not supported
- KDR: Not supported

The Media Gateway should be configured as follows:

- SRTP Preference: SRTP Only
- Cipher Mode: AES Counter Mode
- Authentication Type: SHA1
- Authentication Tag Length: SHA1 32 bits or 80 bits
- Master Key Index (MKI) on Transmit Stream: No
- Key Derivation Enable: No
- Key Derivation Rate (KDR): Not applicable

Example 2:

In this example, the Media Gateway is talking to an IP phone that does not support security:

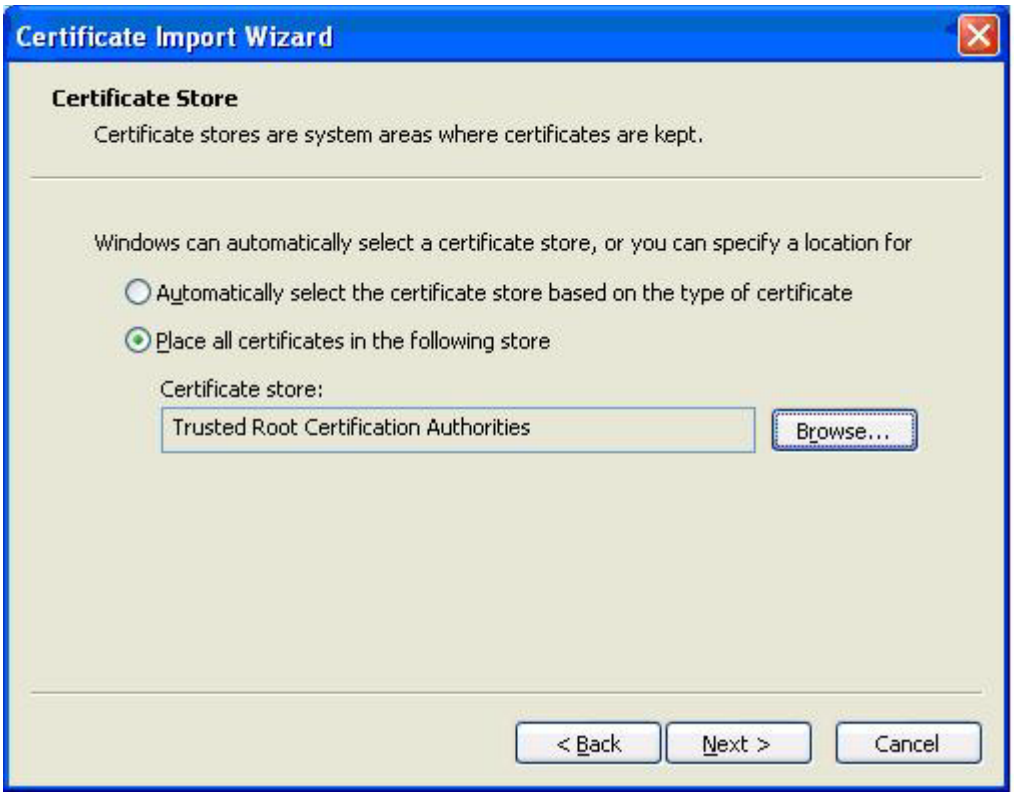
- SRTP Preference: RTP Only
- Master Key Index (MKI) on Transmit Stream: Not applicable
- Key Derivation Enable: Not applicable
- Key Derivation Rate (KDR): Not applicable
- Anti-replay window size hint: Not Applicable
- Cipher Mode: Not applicable
- Authentication Type: Not applicable
- Authentication Tag Length: Not applicable

7.5 Installing Certificate Using Internet Explorer

This section describes the procedure for installing a certificate using Internet Explorer. Perform the following steps:

1. Obtain a copy of the certificate that has already been installed or generated on the Media Gateway.
2. Launch Internet Explorer.
3. Select **Tools > Internet Options…** from the Internet Explorer menu.
4. Select the **Content** tab.
5. Click the **Certificates…** button.
6. Select the **Trusted Root Certificate Authorities** tab.
7. Click the **Import…** button.
8. Click **Next**.
9. Browse to the .cer file.
10. Click **Next**. You should see the following screen:

Figure 45. Storing Self-Signed Certificate by Certificate Import Wizard


Certificate Import Wizard
Certificate Store
Certificate stores are system areas where certificates are kept.
Windows can automatically select a certificate store, or you can specify a location for
Automatically select the certificate store based on the type of certificate
Place all certificates in the following store
Certificate store:
Trusted Root Certification Authorities
Browse...
< Back
Next >
Cancel

11. If the Certificate store is not **Trusted Root Certification Authorities**, use the **Browse** button to locate it.
12. Click **Next**.
13. Click **Finish**.
14. Click **Yes**.
15. Click the **OK** button.
16. Close the windows and Internet Explorer.

Unit Status 8

Status information about the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) units can be obtained through the Web interface.

The DMG1000 Status and DMG2000 Status Web pages allow an administrator to view the number of calls processed by the unit as well as a log of all calls (with call party information). The Media Gateway supports SNMP Version 1. The Media Gateway-specific SNMP MIB can be downloaded from the unit using the Web interface. The MIB file *dmg.mib* can be downloaded from the Media Gateway Status MIB-II Web sub-page.

Types of information available from the Status menu include:

- Summary Information 265
- Alarm Information 266
- Call Log Status Information 266
- Telephony Status Information 267
- MIB-II Status Information 267
- Version Information 268
- Diagnostics Information 268

8.1 Summary Information

Selecting Summary from the Status menu displays the following information:

- Device Summary Information:
  - The Media Gateway MAC address
  - The Media Gateway IP address
  - Uptime of the unit
  - Device Status
  - Ethernet connection losses
  - Telephony carrier losses
  - GateKeeper ID (if applicable)
- Calls Summary Information:
  - Current Calls
  - Inbound IP Calls
  - IP-to-PBX Calls
  - Inbound PBX Calls
  - PBX-to-IP Calls
- Serial Protocol Status (Enabled or Disabled)

8.2 Alarm Information

Selecting Alarm from the Status menu displays the following types of alarm information:

- **Time**- Timestamp of alarm entry.
- **ID** - Identifier of alarm entry.
- **Severity** - Severity Level of alarm entry.
- **Description** - Text Description of alarm entry.
- **Port** - Port Number of alarm entry.
- **Text Message** - Text message of alarm entry.

Clicking on the **Clear Alarms** button will clear all alarm information.

8.3 Call Log Status Information

Selecting Call Log from the Status menu displays information from the Call Log table, including:

- **ID** - Call Record ID number.
- **Start Time** - Displays the starting time of the call. This is the time at which the inbound call rang at the Media Gateway.
- **End Time** - Displays the end time of the call.
- **Source** - A call originates either from the PBX network or the IP network. This field specifies whether the call originated from the PBX network (From Switch Network) or from the IP network (From Packet Network).
- **End Reason** - Displays the reason for ending the call.
- **Inbound Info** - Displays the call party information of the inbound call. If the call is **From Switch Network**, this field contains the call party information of the inbound PBX call. In this case, the format of this field is:

  <PBX port number>:<calling party number>,<calling party name> -> <called party number>, <called party name>

  If the call is **From VoIP Network**, this field contains the call party information of the inbound IP call. In this case, the format of this field is:

  <calling party number>,<calling party name>, <calling party IP> -> <called party number>, <called party name>, <called party IP>
- **Outbound Info** - Displays the call party information of the outbound call. If the call is **From Switch Network**, this field contains the call information used to dial the outbound call to the IP network. In this case, the format of this field is:

  <alias or IP address dialed>

  If the call is **From VoIP Network**, this field contains the call information used to dial the outbound call to the PBX network. In this case, the format of this field is:

  <PBX port number>:<number dialed>

Clicking on the **Clear Log** button will clear all call information from the Call Log table.

8.4 Telephony Status Information

Selecting Telephony from the Status menu displays information about the state of each PBX port that connects to the Media Gateway. For each port, the following status information is provided:

- State (DMG1000):
  - In Service
  - No Link
  - Red Alarm
  - Yellow Alarm
  - Pend D-Chan
  - n/a
- State (DMG2000):
  - In Service
  - No Link
  - Red Alarm
  - Yellow Alarm
  - Pend D-Chan
  - n/a

Note: If the T1/E1 Signaling Mode parameter is set to ISDN, the Port # refers to the T1 or E1 span number.

8.5 MIB-II Status Information

Selecting MIB-II from the Status menu provides a number of Web subpages that include MIB-II information as documented in RFC 1213. The Web subpages include:

- **System** - Provides general Simple Network Management Protocol (SNMP) agent information including: description, object ID, up time, contact information, node name, node location, and the type of services offered.

  From the System Web subpage, you can download the Media Gateway MIB file *dmg.mib.*
- **Interfaces** - Provides Ethernet interface descriptions including: type, speed, physical address, and statistic counters.
- **IP** - Lists counters and address tables for the Internet Protocol (IP) layer.
- **ICMP** - Provides Internet Control Message Protocol (ICMP) packet and error counter information.
- **TCP** - Provides information about the Transmission Control Protocol (TCP) counters and connection table.
- **UDP** - Provides information about the User Datagram Protocol (UDP) counters and local listener table.
- **SNMP** - Records error statistics for SNMP protocol datagrams.

For more information on SNMP Agent, Supported MIBs, and Alarms, refer to the SNMP Application Note at *http://www.dialogic.com/manuals/mediagateway/SNMP_AppNote_6x.pdf*.

8.6 Version Information

Selecting Version from the Status menu displays version information about the Media Gateway software and hardware.

Note: The firmware version information displayed will vary, depending on the Media Gateway model.

Version information includes the following:

- Gateway Application (ROM)
- Gateway Application
- Main Board Boot (ROM)
- DSP Firmware (ROM)
- DSP Firmware
- Telephony Interface Firmware (ROM)
- Telephony Interface Firmware
- Adept Config (ROM)
- Telephony Interface ID
- Port Flags
- Adept Config
- Telephony Interface Application
- Telephony Interface Boot
- Certificate Bundle (ROM)

8.7 Diagnostics Information

Selecting Diagnostics from the Status menu displays diagnostics information about the Media Gateway. Refer to Chapter 9, "Diagnostics" for detailed information about the Media Gateway diagnostic capabilities.

Diagnostics 9

This chapter describes how to perform diagnostics tasks on the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) units and includes the following sections:

- VoIP Interface Test 269
- TDM Interface Test 272
- TDM Self Verification Test 276
- Diagnostic Logging 285
- Communicating to the Terminal Interface 293
- Trace Mechanism 294
- Diagnostic Commands 305

9.1 VoIP Interface Test

VoIP Interface Test is discussed in the following topics:

- VoIP Interface Test Overview
- VoIP Interface Test Operation

9.1.1 VoIP Interface Test Overview

The VoIP interface diagnostic tool is a Web-based user application that can be used to provide verification of VoIP compatibility or troubleshoot the VoIP interface of the Media Gateway.

9.1.1.1 Features

Currently the VoIP interface diagnostic provides the ability to:

- Originate a VoIP call to a specified endpoint address and verify endpoint response.
- Send a message waiting notification status update request and verify endpoint response.

9.1.1.2 Location

The VoIP interface diagnostic tool is only part of the suite of diagnostic utilities that can be used to assist in recording, testing, and resolving configuration or compatibility issues.

The Media Gateway diagnostic utilities are accessible via the Web interface by selecting the *Diagnostics* link on the left side menu of any Web page.

9.1.2 VoIP Interface Test Operation

To operate the VoIP interface diagnostic tool, the user must first navigate to the *Tests-->VoIP* link via the Media Gateway's user interface.

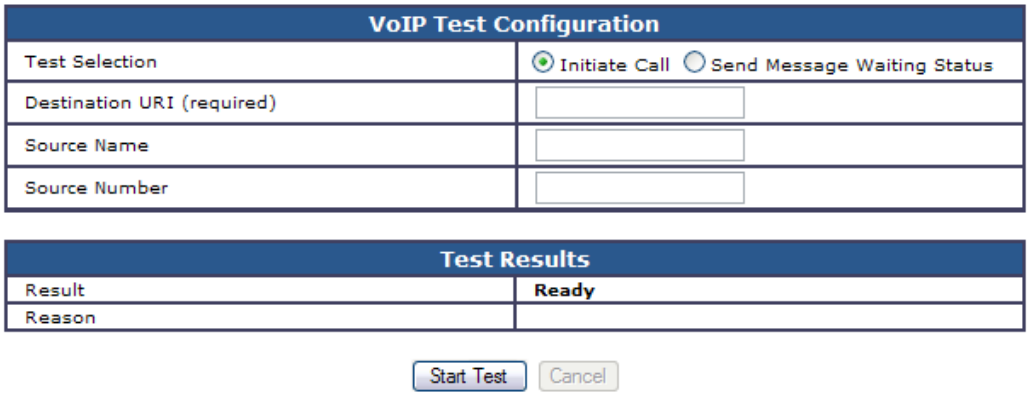
In the VoIP Interface Test Configuration table:

1. Choose the diagnostic test to perform by selecting one of the options in the **Test Selection** field.
2. Then the user must fill in the **Destination VoIP Address** field of the VoIP endpoint that the diagnostic test will use when attempting to establish a connection.
3. Fill in any optional source party information (**Source Name** and **Source Number**) that will be delivered to the destination endpoint during the diagnostic test.
4. If the message waiting diagnostic test is selected, choose the **Messages Waiting Status** to be used in the message waiting notification request.

To start the specified diagnostic test, press the **Start Test** button on the Web page.

Note: Executing the VoIP interface diagnostic tool temporarily disables gateway functionality. Any active connections will be released.

Figure 46. VoIP Interface Web Page



9.1.2.1 Configuration Options

The following configuration items are available to be modified by the user:

Test Selection

Choose the type of diagnostic test to perform. Available options are *Initiate Call* and *Send Message Waiting Status*.

Destination VoIP Address

The **required** address of the VoIP endpoint that the diagnostic test will use when attempting to establish a connection. If this configuration is empty the diagnostic test will not be performed.

Source Name

The alphanumeric name representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Source Number

The number representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Messages Waiting Status

Choose the status value of the message waiting notification update. This field is only valid when the *Message Waiting* diagnostic test is selected.

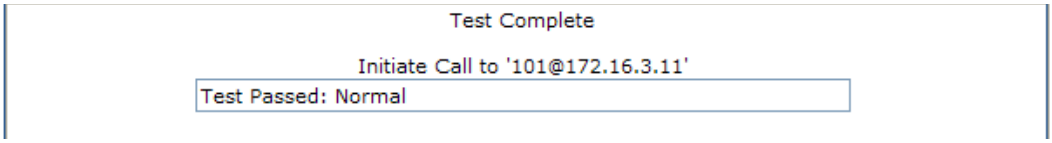
9.1.2.2 Test Status

When a diagnostic test is executed the test results are shown in the VoIP Interface Test Status table on the Web page.

The test can be canceled at any time by pressing the **Cancel Test** button.

When the test is completed a hyperlink will be displayed to allow the user to return to the VoIP Interface Test configuration Web page.

Figure 47. VoIP Interface Test Status Web Page



When the test is completed the results will show if the diagnostic test was able to establish a VoIP connection with the specified **Destination VoIP Address**.

Test Passed: Normal	VoIP connection was successful
Test Failed: *CallEndReason*	VoIP connection failed

The *CallEndReason* description is the error response received either from the specified destination VoIP endpoint or the VoIP interface of the gateway.

Some of the most common error responses and causes are:

Device Not Available	User at VoIP endpoint is not in-service
Transport Failed	VoIP address is invalid or no response received
User Not Found	User not found at VoIP endpoint
Ambiguous Request	VoIP address requires user information

If an error occurs some other suggestions are:

- Verify gateway VoIP interface is in-service and configured properly.
- Verify destination VoIP endpoint is valid and in-service.
- Run the test again with the **Trace Capture** and **Network Capture** diagnostic utilities started to log the detailed call flow of the VoIP interface diagnostic test.

9.1.2.3 Call Log

The test results from the VoIP interface diagnostic tool are also stored in the Media Gateway call log. The *Source* field will contain **From Test App** to indicate that the connection was created by a diagnostic application.

The call log is accessible through the Web by selecting the *Call Log* link on the left side menu of any Web page.

Figure 48. VoIP Interface Call Log Web Page

ID	Start Time	End Time	Source	End Reason	Inbound info	Outbound info
3	2/16 9:52:00	2/16 9:52:01	From Test App	VoIP: Normal	Mwi Clear 1234,John Doe->,	,,101@172.16.3.11,
2	2/16 9:51:52	2/16 9:51:52	From Test App	VoIP: Normal	Mwi Set 1234,John Doe->,	,,101@172.16.3.11,
1	2/16 9:51:43	2/16 9:51:43	From Test App	VoIP: Normal	1234,John Doe->,->, [Rsn=Direct]	,,101@172.16.3.11,

9.2 TDM Interface Test

TDM Interface Test is discussed in the following topics:

- TDM Interface Test Overview
- TDM Interface Test Operation

9.2.1 TDM Interface Test Overview

The TDM interface diagnostic tool is a Web-based user application that can be used to provide verification of TDM compatibility or troubleshoot the TDM interface of the Media Gateway.

9.2.1.1 Features

Currently the TDM interface diagnostic provides the ability to:

- Originate a TDM call to a specified endpoint address and verify endpoint response.
- Send a message waiting notification status update request and verify endpoint response.

9.2.1.2 Location

The TDM interface diagnostic tool is only part of the suite of diagnostic utilities that can be used to assist in recording, testing, and resolving configuration or compatibility issues.

The Media Gateway diagnostic utilities are accessible via the Web interface by selecting the *Diagnostics* link on the left side menu of any Web page.

9.2.2 TDM Interface Test Operation

To operate the TDM interface diagnostic tool, the user must first navigate to the *Tests-->TDM* link via the Media Gateway's user interface.

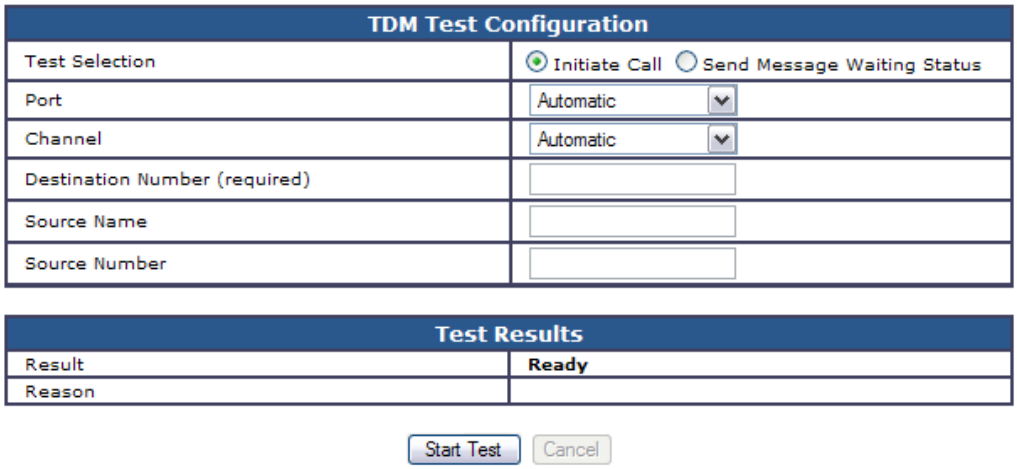
In the TDM Interface Test Configuration table:

1. Choose the diagnostic test to perform by selecting one of the options in the **Test Selection** field.
2. The user must fill in the **Destination Number** field of the TDM endpoint that the diagnostic test will use when attempting to establish a connection.
3. Optionally, select the interface and channel index of the gateway to use in the diagnostic test by choosing the **Interface Selection** or **Channel Selection**. If *Automatic* is chosen the test will select the first interface and channel available.
4. Fill in any optional source party information (**Source Name** and **Source Number**) that will be delivered to the destination endpoint during the diagnostic test.
5. If the message waiting diagnostic test is selected, choose the **Messages Waiting Status** to be used in the message waiting notification request.

To start the specified diagnostic test, press the **Start Test** button on the Web page.

Note: Executing the TDM interface diagnostic tool temporarily disables gateway functionality. Any active connections will be released.

Figure 49. TDM Interface Web Page



9.2.2.1 Configuration Options

The following configuration items are available to be modified by the user:

Test Selection
Choose the type of diagnostic test to perform. Available options are *Initiate Call* and *Send Message Waiting Status*.

Interface Selection
Select the specific interface to use in the diagnostic test. If a specific interface is not chosen the test will select the first interface available.

Channel Selection
Select the specific interface to use in the diagnostic test. If a specific interface is not chosen the test will select the first interface available. This option is only available on TDM interfaces that have more than one bearer channel.

Destination Number
The **required** number of the TDM endpoint that the diagnostic test will use when attempting to establish a connection. If this configuration is empty the diagnostic test will not be performed.

Source Name
The alphanumeric name representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Source Number
The number representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Messages Waiting Status
Choose the status value of the message waiting notification update. This field is only valid when the *Message Waiting* diagnostic test is selected.

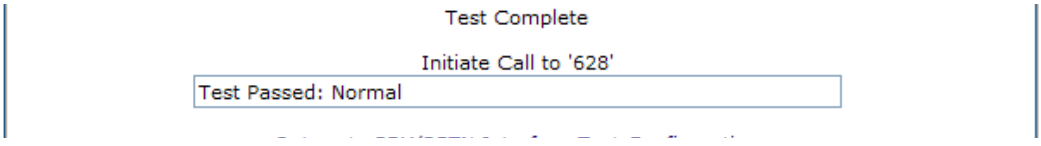
9.2.2.2 Test Status

When a diagnostic test is executed the test results are shown in the TDM Interface Test Status table on the Web page.

The test can be canceled at any time by pressing the **Cancel Test** button.

When the test is completed a hyperlink will be displayed to allow the user to return to the TDM Interface Test configuration Web page.

Figure 50. TDM Interface Test Status Web Page



When the test is completed the results will show if the diagnostic test was able to establish a TDM connection with the specified **Destination Number**.

Test Passed: Normal	TDM connection was successful
Test Failed: *CallEndReason*	TDM connection failed

The *CallEndReason* description is the error response received either from the specified destination TDM endpoint or the TDM interface of the gateway.

Some of the most common error responses and causes are:

Device Not Available	TDM interface is not in-service
User Busy	Destination number is busy (in-use)
User Not Found	Destination number not found at TDM endpoint

If an error occurs some other suggestions are:

- Verify gateway TDM interface is in-service and configured properly.
- Verify destination TDM endpoint is valid and in-service.
- Run the test again with the **Trace Capture** and **Network Capture** diagnostic utilizes started to log the detailed call flow of the VoIP interface diagnostic test.

9.2.2.3 Call Log

The test results from the TDM interface diagnostic tool are also stored in the Media Gateway call log. The *Source* field will contain **From Test App** to indicate that the connection was created by a diagnostic application.

The call log is accessible through the Web by selecting the *Call Log* link on the left side menu of any Web page.

Figure 51. TDM Interface Call Log Web Page

ID	Start Time	End Time	Source	End Reason	Inbound info	Outbound info
3	2/16 10:10:42	2/16 10:10:42	From Test App	TDM: Normal	Mwi Clear 1234,John Doe->628,	1:628
2	2/16 10:10:35	2/16 10:10:35	From Test App	TDM: Normal	Mwi Set 1234,John Doe->628,	1:628
1	2/16 10:10:25	2/16 10:10:26	From Test App	TDM: Normal	1234,John Doe->628,->, [Rsn=Direct]	1:1 628

9.3 TDM Self Verification Test

TDM Self Verification is discussed in the following topics:

- TDM Self Verification Test Overview
- TDM Self Verification Test Operation

9.3.1 TDM Self Verification Test Overview

The TDM self verification diagnostic tool is a Web page user application that can be used to provide verification of PBX compatibility and also to confirm the Media Gateway configuration values relating to the PBX interface are valid.

9.3.1.1 Features

Currently the TDM self verification diagnostic provides the ability to verify the following operations:

Initiate Call

- Originate an outbound PBX call
- Receive outbound call progress (alerting)
- Send and Receive DTMF digits
- Receive far-end disconnect supervision

Answer Call

- Answer an incoming PBX call
- Receive call party identification (CPID)
- Send and Receive DTMF digits
- Release an active call

Transfer Call

- Transfer a connected PBX call
- Answer an incoming (transferred) PBX call
- Receive call party identification (CPID) during incoming transfer
- Receive DTMF digits after transfer completed

Message Waiting

- Send message waiting notification request (Set)
- Send no message waiting notification request (Clear)

9.3.1.2 Restrictions

The TDM self verification diagnostic is currently only available on PBX interfaces and protocols that allow the interfaces and channels to be individually assigned extension numbers and allows outbound and inbound call routing from one interface and channel to another.

Note: The TDM self verification diagnostics is not available on any ISDN protocol.

9.3.1.3 Capabilities

The ability to route calls and send message waiting status updates through a particular PBX interface and channel is determined by the capabilities, enabled state, and if the interface and channel is in-service. If the interface and channel is not in-service or is not enabled, or it does not support the capability for the requested action, then the interface and channel will not be used for routing through the gateway and will not be included in a diagnostic test.

To view and modify the capabilities and enabled state on a PBX interface and channel navigate to the *Gateway -> Gateway Capabilities* Web page on the Media Gateway.

9.3.1.4 Location

The TDM self verification diagnostic tool is only part of the suite of diagnostic utilities that can be used to assist in recording, testing, and resolving configuration or compatibility issues.

The Media Gateway diagnostic utilities are accessible via the Web interface by selecting the *Diagnostics* link on the left side menu of any Web page.

9.3.2 TDM Self Verification Test Operation

To operate the TDM self verification diagnostic tool, the user must first navigate to the *Tests-->TDM Self Test* link via the Media Gateway's user interface.

In the TDM Self Verification Test Configuration table:

1. Choose the diagnostic test to perform by selecting one of the options in the **Test Selection** field. The default selection is the *Initiate Call / Answer Call* diagnostic test only, but all diagnostic test calls can be selected to be executed as one operation.
2. If the *Initiate Call / Answer Call* or *Transfer Call* diagnostic tests are selected, fill in the interface extension numbers of the interfaces that are to be included in the test.
   - Initiate Call / Answer Call - Requires two valid interface extension numbers.
   - Transfer Call - Requires three valid interface extension numbers and is executed with the Initiate Call / Answer Call diagnostic test.
3. If the Send Message Waiting Status diagnostic test is selected, fill in the **Message Waiting Extension Number** field with the extension to send the status update requests.

Note: The *Send Message Waiting Status* diagnostic test does not allow individual interfaces to be selected for inclusion/exclusion. The test will be executed on all available PBX interfaces.

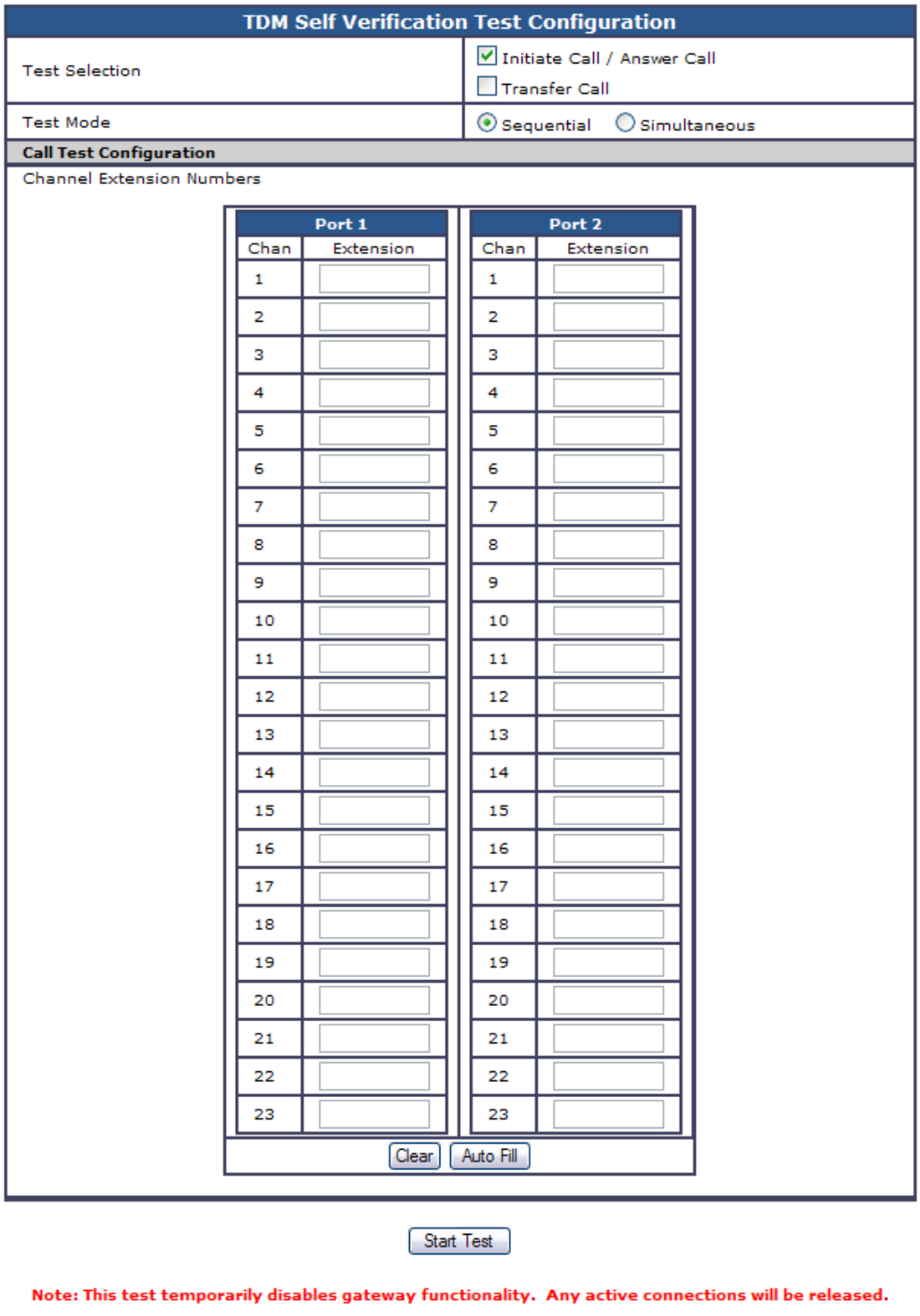
4. Select the **Test Mode** to be *Sequential* or *Simultaneous* (this is an optional parameter for the test).

Note: The *Send Message Waiting Status* diagnostic tests are always executed in Sequential mode due to possible load issues of the PBX under test.

To start the specified diagnostic test, press the **Start Test** button on the Web page.

Note: Executing the TDM interface diagnostic tool temporarily disables gateway functionality. Any active connections will be released.

Figure 52. TDM Self Verification Web Page



9.3.2.1 Configuration Options

The following configuration items are available to be modified by the user:

Interface Extension Numbers

Enter the extension number of the interfaces to include in the selected diagnostic test. Interface extension numbers are only valid for *Initiate Call / Answer Call*, and *Transfer Call* diagnostic tests. Interface extension numbers that are left blank will not be included in the diagnostic test.

Message Waiting Extension Number

Enter the extension number to send the message waiting status update requests. Message waiting extension number is only valid for the *Send Message Waiting Status* diagnostic test selection.

Test Selection

Choose the type of diagnostic test to perform. Available options are *Initiate Call / Answer Call, Transfer Call* and *Send Message Waiting Status*.

Test Mode

Select the execution mode of the diagnostic test to be either *Simultaneous* or *Sequential*. Sequential test mode executes only a single diagnostic test at one time. The next diagnostic test will not start until the previous test has completed. Simultaneous test mode executes as many diagnostic tests at once as possible. The next diagnostic test will start as soon as the previous test has started.

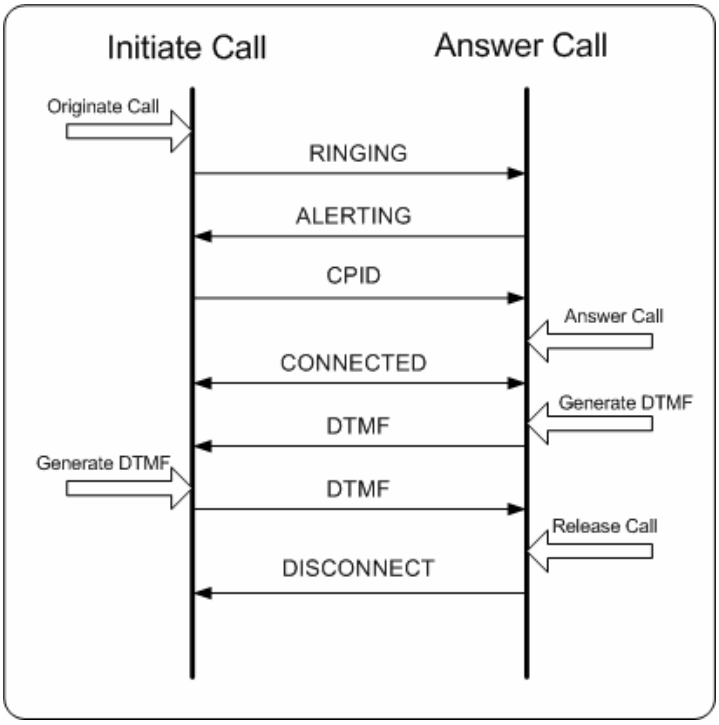
The *Send Message Waiting Status* diagnostic tests are always executed in Sequential mode due to possible load issues of the PBX under test.

9.3.2.2 Call Flows

Initiate Call / Answer Call

The call flow of the *Initiate Call / Answer Call* diagnostic test tries to verify the operation of originating an outbound call and answering an inbound call. These tests also include verification of outbound call progress, inbound CPID, send/receive DTMF and disconnect supervision.

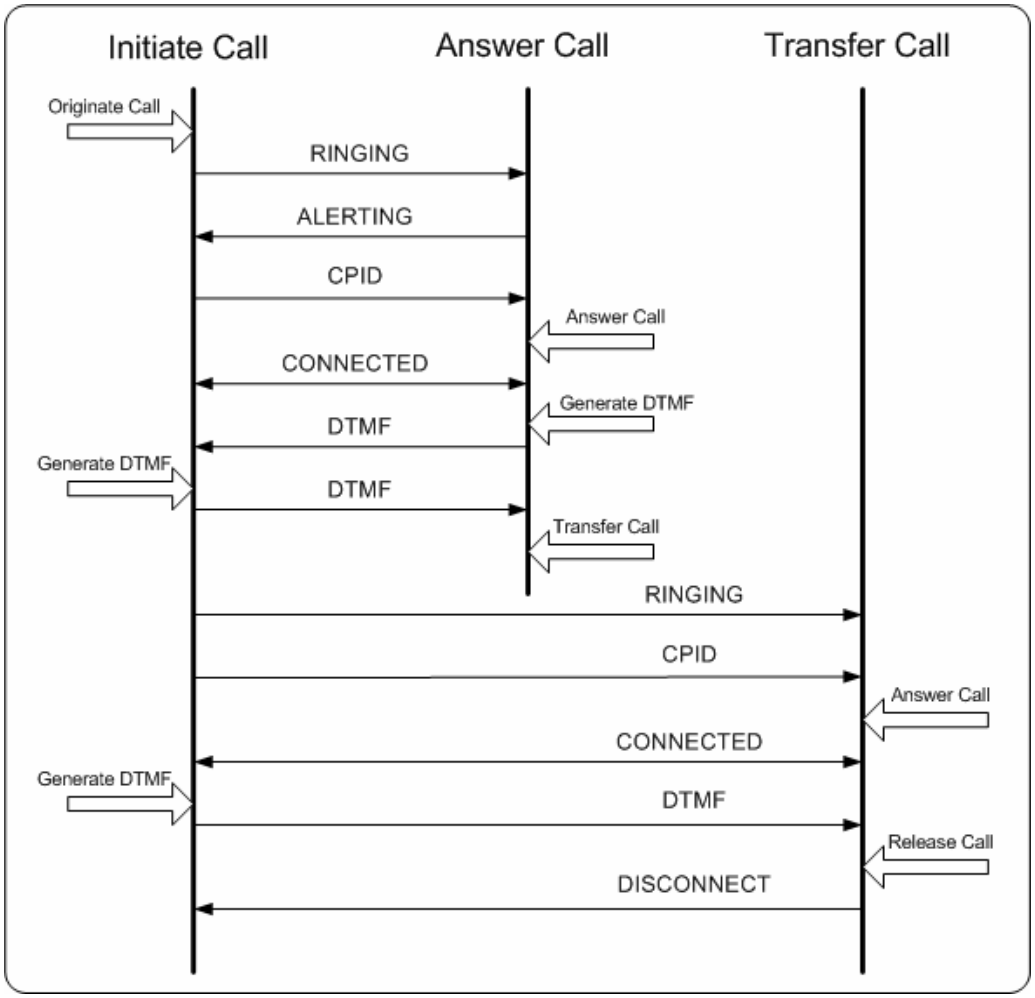
Figure 53. Call Flow for Initiate Call / Answer Call



Initiate Call / Answer Call and Transfer Call

The call flow of the *Initiate Call* / *Answer Call* and *Transfer Call* diagnostic test tries to verify not only the operation of originating an outbound call and answering an inbound call, but transferring the call as well. These tests also include verification of outbound call progress, inbound CPID, send/receive DTMF and disconnect supervision.

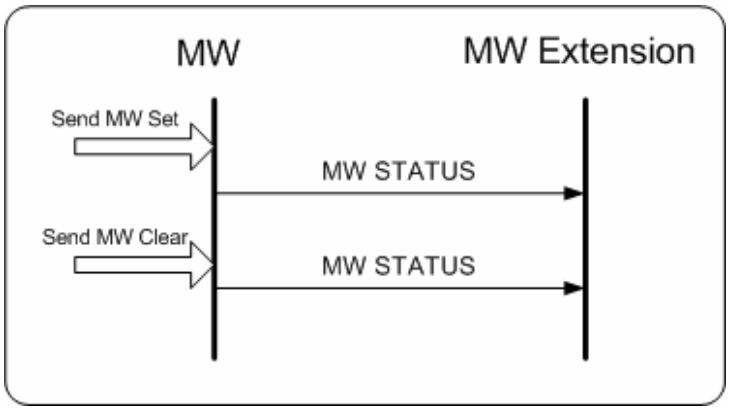
Figure 54. Call Flow for Initiate Call / Answer Call and Transfer Call



Send Message Waiting Status

The call flow of the *Send Message Waiting Status* diagnostic test tries to verify the message waiting notification set and clear update operation.

Figure 55. Call Flow for Send Message Waiting Status



9.3.2.3 Test Status

When a diagnostic test is executed the test results are shown in the TDM Self Verification Test Status table on the Web page.

Test Result Symbol	Description
-	Test skipped because not configured or not available (no capability, not enabled, or not in-service).
n	Test was pending but did not execute because a failure occurred.
P	Test Passed
F	Test Failed

The test can be canceled at any time by pressing the **Cancel Test** button.

When the test is completed a hyperlink will be displayed to allow the user to return to the TDM Self Verification Test configuration Web page.

Figure 56. TDM Self Verification Test Status Web Page

Intfc	Status	Outbound Route	Initiate Call				Answer Call				Transfer Call				MW	
			Orig	Prog	DTMF	Disc	Ans	CPID	DTMF	Xfr	Ans	CPID	DTMF	Rel	Set	Clr
1	Answer	1:2000->2:2001->4:2003	P	P	P	P	n	P	n	n	P	P	P	P	-	-
2	Transfer	2:2001->4:2003->1:2000	P	P	P	P	P	P	P	P	n	n	n	n	-	-
3	Idle	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-
4	Initiate	4:2003->1:2000->2:2001	P	P	n	n	P	P	P	P	P	P	P	P	-	-
5	Answer	5:2004->6:2005->8:2007	P	P	P	P	n	P	n	n	P	P	P	P	-	-
6	Transfer	6:2005->8:2007->5:2004	P	P	P	P	P	P	P	P	n	n	n	n	-	-
7	Idle	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-
8	Initiate	8:2007->5:2004->6:2005	P	P	n	n	P	P	P	P	P	P	P	P	-	-

9.3.2.4 Test Results

After the diagnostic test(s) are completed or canceled the test results can be opened or saved to a text file (.txt) for documented proof of the verification process. To open the test results in another Web browser window click the ***pbxtest.txt*** hyperlink.

Figure 57. TDM Self Verification Test Results

```
Test Selection: Initiate Call|Answer Call|Transfer Call|MW
Test Mode:      Simultaneous
MW Extension:   2008

Intfc  Status         Outbound Route         |   Initiate Call   |    Answer Call    |   Transfer Call   |    MW   |
                                             |Orig Prog DTMF Disc|Ans  CPID DTMF Xfr |Ans  CPID DTMF Rel |Set  Clr |
 1    Complete  1:2000->2:2001->4:2003         P    P    P    P    P    P    P    P    P    P    P    P    P    P
 2    Complete  2:2001->4:2003->1:2000         P    P    P    P    P    P    P    P    P    P    P    P    P    P
 3    Complete                                 -    -    -    -    -    -    -    -    -    -    -    -    -    -
 4    Complete  4:2003->1:2000->2:2001         P    P    P    P    P    P    P    P    P    P    P    P    P    P
 5    Complete                                 -    -    -    -    -    -    -    -    -    -    -    -    P    P
 6    Complete                                 -    -    -    -    -    -    -    -    -    -    -    -    P    P
 7    Complete                                 -    -    -    -    -    -    -    -    -    -    -    -    -    -
 8    Complete                                 -    -    -    -    -    -    -    -    -    -    -    -    P    P


Test Result Symbols
-------------------
- = results n/a
n = not tested
P = passed
F = failed

Initiate Call Test Descriptions
-----------------------------------------------
Orig - Originate an outbound call
Prog - Receive outbound call progress (alerting)
DTMF - Send\Receive DTMF digits
Disc - Receive far-end disconnect supervision

Answer Call Test Descriptions
-----------------------------------------------
Ans  - Answer an incoming call
CPID - Receive call party identification
DTMF - Send\Receive DTMF digits
Xfr  - Transfer call

Transfer Call Test Descriptions
-----------------------------------------------
Ans  - Answer an incoming call
CPID - Receive call party identification
DTMF - Receive DTMF digits
Rel  - Release an active call

Message Waiting Test Descriptions
-----------------------------------------------
Set  - Send messages waiting notification
Clr  - Send no message waiting notification
```

9.3.2.5 Test Failures

If a failure occurs some suggestions are:

- Verify PBX interface is in-service and configured properly
- Verify gateway is configured for specific PBX interface (is there a default .ini file)
- Run the test again with a minimum number of extensions configured and in *Sequential* test mode.
- Run the test again with the **Trace Capture** diagnostic utility started to log the detailed call flow of the TDM self verification diagnostic test.
- Use the **TDM Interface** diagnostic utility to test *Initiate Call* and *Send Message Waiting Status* on a particular interface and channel.

- Check the Call Log for the call end reason of the connections used in the diagnostic tests.

Some of the most common test result failures and causes are:

Test	Failure Reasons
Originate an outbound PBX call	• The extension numbers for the PBX interface are not correct. • The PBX interface and channel are not configured properly*. • Dial tone is not detected or being provided.
Answer an outbound PBX call	• The extension numbers for the PBX interface are not correct. • The PBX interface and channel are not configured properly*.
Receive outbound call progress (alerting)	• Call progress is not detected or being provided.
Receive call party identification (CPID)	• The switch is not providing CPID. • The programmed CPID type on the Media Gateway does not match the CPID type being provided by the switch (ex. Type I or Type II DMTF)
Send and Receive DTMF digits	• The PBX interface and channel are not configured properly*. • The switch does not allow DTMF between interfaces and channels.
Receive far-end disconnect supervision	• The PBX interface and channel are not configured properly*. • Disconnect tone is not detected or being provided.
Transfer a connected PBX call	• The extension numbers for the PBX interface are not correct. • The PBX interface and channel are not configured properly*. • Dial tone is not detected or being provided.
Send messages waiting notification request	• The PBX interface and channel are not configured properly*. • No confirmation tone from the switch. • Feature Access Codes (FAC) is not correct. • The MWI extension number is not correct or does not accept message waiting status updates. • The MWI uses serial port and/or it is not configured properly.
*For more details on configuration, see the *Installation and Configuration Integration Notes*.	

9.3.2.6 Call Log

The test results from the TDM self verification diagnostic tool are also stored in the Media Gateway call log. The *Source* field will contain **From Test App** to indicate that the connection was created by a diagnostic application.

The call log is accessible through the Web by selecting the *Call Log* link on the left side menu of any Web page.

Figure 58. TDM Self Verification Call Log Web Page

ID	Start Time	End Time	Source	End Reason	Inbound info	Outbound info
21	2/21 9:13:00	2/21 9:13:02	From Test App	TDM: Normal	Mwi Clear ,->2008,	8:2008
20	2/21 9:12:59	2/21 9:13:00	From Test App	TDM: Normal	Mwi Set ,->2008,	8:2008
19	2/21 9:12:57	2/21 9:12:59	From Test App	TDM: Normal	Mwi Clear ,->2008,	6:2008
18	2/21 9:12:56	2/21 9:12:57	From Test App	TDM: Normal	Mwi Set ,->2008,	6:2008
17	2/21 9:12:55	2/21 9:12:56	From Test App	TDM: Normal	Mwi Clear ,->2008,	5:2008
16	2/21 9:12:53	2/21 9:12:55	From Test App	TDM: Normal	Mwi Set ,->2008,	5:2008
15	2/21 9:12:52	2/21 9:12:53	From Test App	TDM: Normal	Mwi Clear 2003,->2008,	4:2008
14	2/21 9:12:50	2/21 9:12:52	From Test App	TDM: Normal	Mwi Set 2003,->2008,	4:2008
13	2/21 9:12:49	2/21 9:12:50	From Test App	TDM: Normal	Mwi Clear 2001,->2008,	2:2008
12	2/21 9:12:48	2/21 9:12:49	From Test App	TDM: Normal	Mwi Set 2001,->2008,	2:2008
11	2/21 9:12:46	2/21 9:12:48	From Test App	TDM: Normal	Mwi Clear 2000,->2008,	1:2008
10	2/21 9:12:45	2/21 9:12:46	From Test App	TDM: Normal	Mwi Set 2000,->2008,	1:2008
9	2/21 9:12:43	2/21 9:12:45	From TDM Network	TDM: Normal	2:2003,->,->, [Rsn=Direct]	
8	2/21 9:12:33	2/21 9:12:45	From TDM Network	TDM: Normal	1:2003,->,->, [Rsn=Direct]	
7	2/21 9:12:31	2/21 9:12:45	From Test App	TDM: Normal	2003,->2000,->, [Rsn=Direct]	4:1 2000
6	2/21 9:12:29	2/21 9:12:31	From TDM Network	TDM: Normal	1:2001,->,->, [Rsn=Direct]	
5	2/21 9:12:19	2/21 9:12:31	From TDM Network	TDM: Normal	4:2001,->,->, [Rsn=Direct]	
4	2/21 9:12:18	2/21 9:12:31	From Test App	TDM: Normal	2001,->2003,->, [Rsn=Direct]	2:1 2003
3	2/21 9:12:15	2/21 9:12:17	From TDM Network	TDM: Normal	4:2000,->,->, [Rsn=Direct]	
2	2/21 9:12:05	2/21 9:12:17	From TDM Network	TDM: Normal	2:2000,->,->, [Rsn=Direct]	
1	2/21 9:12:04	2/21 9:12:17	From Test App	TDM: Normal	2000,->2001,->, [Rsn=Direct]	1:1 2001

9.4 Diagnostic Logging

The following is an overview describing the diagnostic logging capabilities of the Media Gateway and includes the following sections:

- Overview
- Trace Capture
- Network Capture
- TDM Capture

9.4.1 Overview

The Media Gateway provides the user with the ability to record, in real-time, diagnostic information. This information can then be transferred to a PC for off-line analysis of problems.

Three independent logs are available:

- **Trace Capture:** The Trace log is a verbatim copy of the trace messages that are available using the terminal interface. The messages are described in Section 9.6, “Trace Mechanism”, on page 294.
- **Network Capture:** The network log is a copy of the IP messages transmitted and received by the Gateway.

- **TDM Capture:** The TDM log is a copy of the voice data transferred on the T1/E1 line. It is available only on the DMG2000.

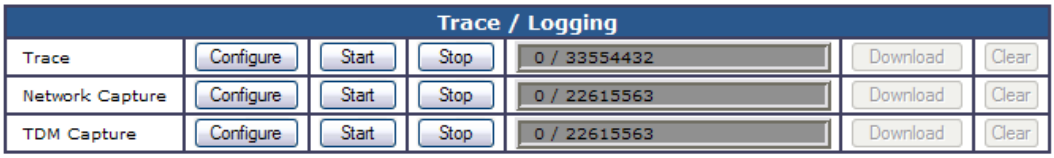
To view a log you must transfer the log to a PC. When transferred to a PC, each log is formatted into an industry standard to facilitate viewing and sharing of data. The format for each log is:

- **Trace Capture**: Uses an ASCII text format (.TXT)
- **Network Capture**: Uses an Ethereal format. Ethereal is a freeware program downloadable from the Internet. It is required only to view Network Captures, not to transfer them from the Media Gateway to a PC.
- **TDM Capture**: Uses a Wave file format (.wav). Wave format is a standard Windows® format and can be opened by programs such as Windows Media Player, Cool Edit and Adobe Audition. None of these programs are required to transfer the file from the Gateway to a PC, but one is required to view or playback the file.

9.4.1.1 Control of Log Files:

The control of each log file is independent (i.e. more than one log can be simultaneously created) and is administered via the Media Gateway's Web interface. You reach the logging pages by using the Diagnostic Web page, shown in Figure 59.

Figure 59. Diagnostic Web Page



In summarizing the overview of Diagnostic Logging, remember:

- Logging is available on Media Gateway Version 5.0 Software and later.
- The DMG1000 provides two logs: Trace capture and Network capture.
- The DMG2000 provides three logs: Trace capture, Network capture and TDM capture.
- The Media Gateway stores the logs in RAM. Therefore, the logs are not preserved if the unit is reset or powered down. To save a log you must transfer it to a PC.
- Special applications are not required to download logs from the Media Gateway to a PC, but are required to view the logs.
- All control of logging is through the Web interface.

9.4.2 Trace Capture

The Trace Capture enhances the trace mechanism described in Section 9.6, "Trace Mechanism", on page 294. Specifically, it allows users to set the trace mask via the Web page in addition to allowing the trace output to be redirected to a log file that can then be downloaded. The DMG2000 Trace

Capture control page is shown in Figure 60. While the DMG2000 control page has more trace keys, the ideas presented in this section apply equally to both the DMG1000 and the DMG2000.

Figure 60. Trace Capture Control Page - DMG2000

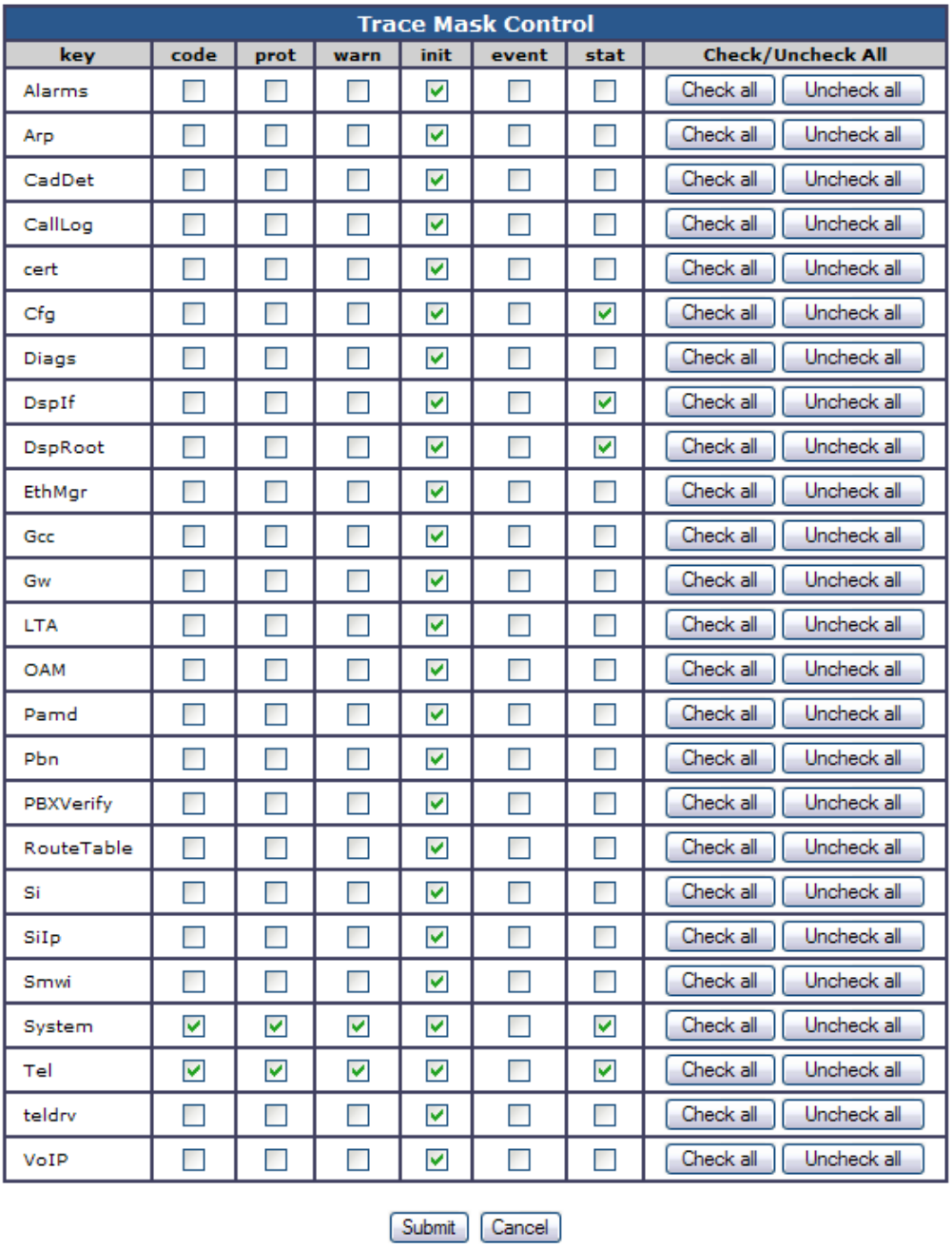
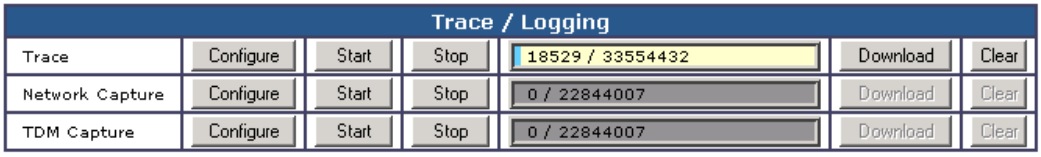


Figure 60 shows the default trace keys (i.e. immediately after power-up). Notice the Error and Trace masks are not shown as they are always enabled.

To use the Trace Capture, first check the appropriate trace masks. Click **Apply Masks** and then click **Start Logging**. Logging starts and the display changes to show the state of the log. An example of a running log is shown in Figure 61.

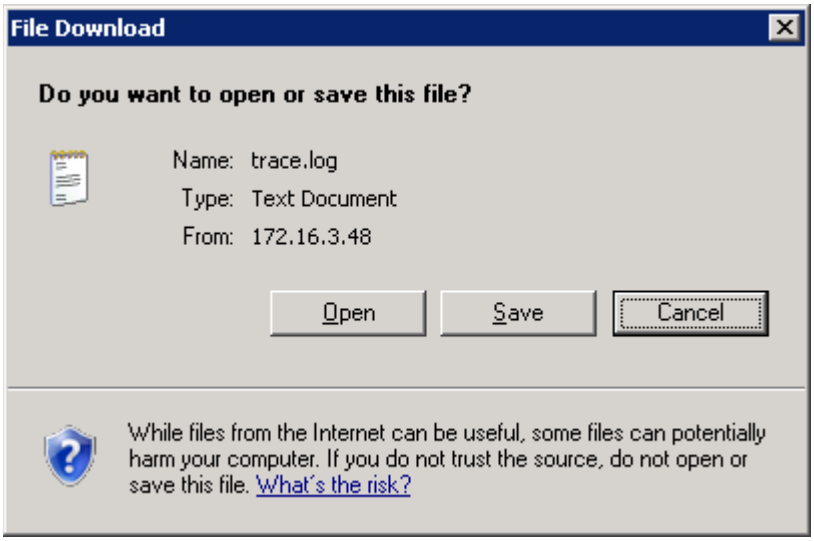
Figure 61. Example of a Running Log - Trace Capture



When the Trace Capture is running, the fill level of the log is displayed. The log uses a circular buffer, so when it reaches capacity, new data overwrites the old. Once the log reaches capacity, the fill level does not change.

The log can be transferred to a PC by clicking **Download**. Doing this brings up a File Download dialog box as shown in Figure 62. Select **Save** to save the file to your PC. After saving, you can view the file using most any ASCII text editor, such as Wordpad or Notepad.

Figure 62. File Download Dialog Box for Trace.log


File Download
Do you want to open or save this file?
Name: trace.log
Type: Text Document
From: 172.16.3.48
Open
Save
Cancel
While files from the Internet can be useful, some files can potentially harm your computer. If you do not trust the source, do not open or save this file. What's the risk?

Important points to remember about Trace Capture are:

- After modifying the trace keys be sure to click Apply Masks.
- It is not necessary to stop logging to change trace keys. The keys can be changed on the fly.
- Downloading the Trace Capture clears the log and starts a new log file.
- Stopping the log and then restarting it does not clear the log, new data is just appended to the log.

- The DMG1000 log can store up to 1.5 Mbytes, while the DMG2000 log can store up to 32 Mbytes.
- Trace Capture is automatically enabled at power-on-with Error, Debug and Init trace masks for all trace keys.

9.4.3 Network Capture

The Network Capture creates an in-memory log of Media Gateway IP traffic. Since there can be a great deal of traffic, the Network Capture control page allows filtering of IP packets. Filtering packets makes it easier to analyze problems by reducing the number of irrelevant messages. It also increases the time before the log wraps, thus making it more probable an error will be caught.

9.4.3.1 Network Capture Details

The DMG1000 unit has one network interface:

- LAN1: an external network that connects the DMG1000 to a VOIP network

The DMG2000 unit has three network interfaces:

- DSP: an internal network that connects the DMG2000 DSP to the DMG2000 CPU
- LAN1: an external network that connects the DMG2000 to a network (VOIP)
- LAN2: an external network that connects the DMG2000 to a network (non-VOIP)

Notes: ***1.*** The DMG1000's LAN1 is equivalent to the DMG2000's LAN1.

2. Currently, LAN2 is only supported in Version 5.1 SU1 Software or later.

The Network Capture is controlled from the Media Gateway's Web interface. The DMG2000 Network Capture control page is shown in Figure 63. Even though the DMG1000 control page has only one network interface (LAN1), the ideas presented in this section apply equally to the DMG2000 and the DMG1000.

Figure 63. Network Capture Control Page - DMG2000

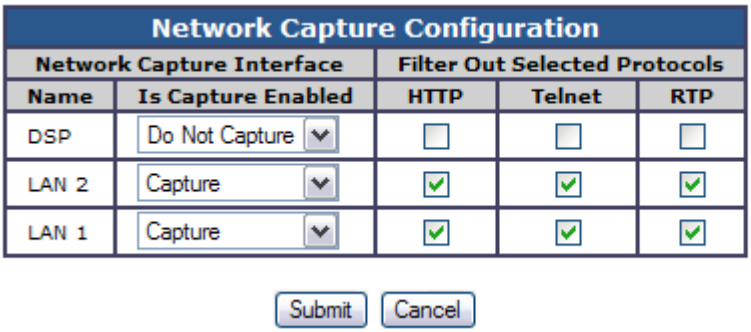
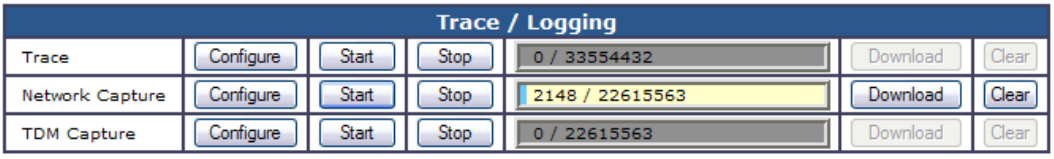


Figure 63 shows the default values (i.e. immediately after power-up) for a DMG2000. The first level of filtering occurs on the interface level. You can select **Capture** to enable or **Do not capture**

to disable individual interfaces. The second level of filtering is on the protocol level. When an interface is enabled (Capture), you have the ability to filter (not log) different protocols. A check mark in the box means the Gateway will not log packets of the corresponding protocol.

To use the Network Capture, first select the interfaces to capture, check the protocols you do not want to log, click **Apply Filters** and then click **Start Capturing**. Logging starts and the display changes to show the state of the log. An example of a running log is shown in Figure 64.

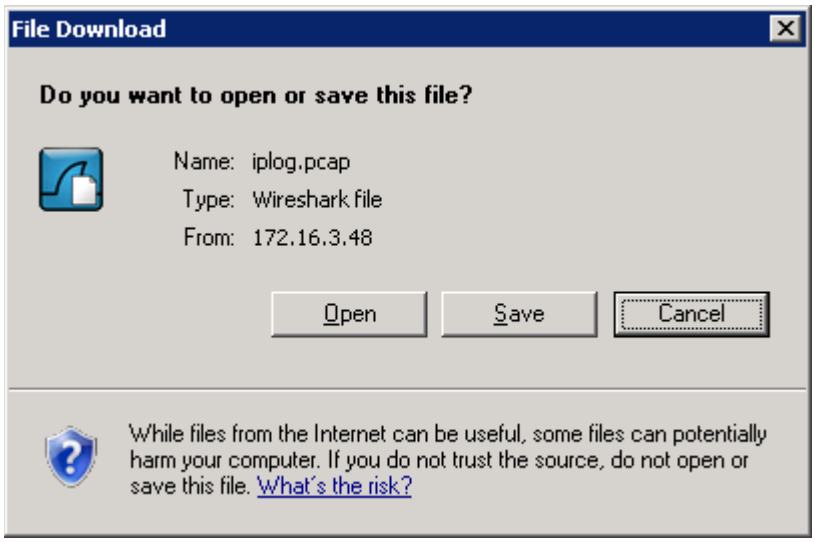
Figure 64. Example of Running Log - Network Capture



When the Network Capture is running, the fill level of the log is displayed. The log uses a circular buffer, so when it reaches capacity, new data overwrites the old. Once the log reaches capacity the fill level does not change.

The log can be transferred to a PC by clicking **Download**. Doing this brings up a File Download dialog box as shown in Figure 65. Select **Save** to save the file to the PC. After saving, you can view the file with any network capture program that supports libpcap format (e.g. Ethereal).

Figure 65. File Download Dialog Box for Iplog.pcap


File Download
Do you want to open or save this file?
Name: iplog.pcap
Type: Wireshark file
From: 172.16.3.48
Open
Save
Cancel
While files from the Internet can be useful, some files can potentially harm your computer. If you do not trust the source, do not open or save this file. What's the risk?

Important points to remember about Network Capture are:

- Downloading does not require any special software, but viewing requires Ethereal.
- On the Network Capture Web page, checking a protocol type causes it NOT to be captured.

- After selecting protocol filters, be sure to click Apply Filters.
- Size of the log is determined at startup. DMG1000 ~1.1 Mbytes, DMG2000 ~24 Mbytes.
- Downloading the Network Capture clears the log and starts another.
- Stopping the log and restarting it does not clear the log, new data is just appended.

9.4.4 TDM Capture

The TDM Capture creates an in-memory log of voice traffic on the DMG2000's T1 or E1 line. The capture is controlled from the DMG2000's Web interface, as shown in Figure 66. You can log voice traffic from a single channel in either the Rx direction, or the Rx and Tx direction, where Rx is defined as entering the DMG2000 and Tx as leaving the DMG2000. The TDM Capture is not supported by the DMG1000.

Figure 66. TDM Capture Control Web Page

9.4.4.1 TDM Capture Details

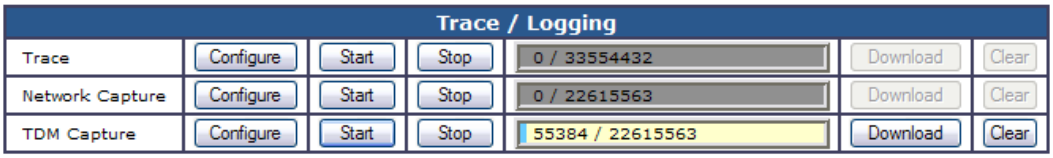
Figure 66 shows the default values (i.e. immediately after power-up) for the TDM Capture. You can select any one channel for logging. The range of channels is dependent on the TDM configuration (T1 or E1) and the number of hardware spans. Table 18 shows the mapping from a protocol and span to the TDM Capture channel numbers.

Table 18. Mapping of Protocol and Span Numbers to TDM Capture Channel Numbers

	Span 1	Span 2	Span 3	Span 4
T1 CAS	1 to 24	25 to 48	49 to 72	73 to 96
T1 QSIG	1 to 23	24 to 46	47 to 69	70 to 92
T1 NI-2	1 to 23	24 to 46	47 to 69	70 to 92
E1 QSIG	1 to 30	31 to 60	61 to 90	91 to 120

To use the TDM Capture, enter the Channel Number to capture, select either RX or RX & TX, then click **Start Capturing**. Logging begins and the display changes to show the state of the log. An example of a running log is shown in Figure 67.

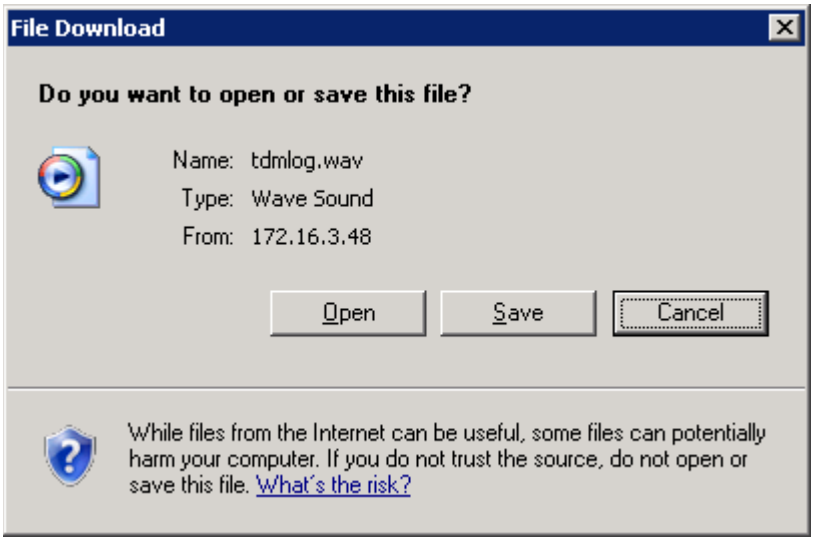
Figure 67. Example of Running Log - TDM Capture



While the TDM Capture is running, the fill level of the log is displayed. The log is a circular buffer, so when it reaches capacity, new data overwrites the old. Once the log reaches capacity, the fill level does not change.

The log can be transferred to a PC clicking on **Download**. Doing this brings up a File Download dialog box as shown in Figure 68. Select Save to save the file to the PC. After saving, you can play the file using Windows Media Player or view it with a variety of audio file viewers (ex. CoolEdit Pro, Adobe Audition).

Figure 68. File Download Dialog Box for Tdmlog.wav


File Download
Do you want to open or save this file?
Name: tdmlog.wav
Type: Wave Sound
From: 172.16.3.48
Open
Save
Cancel
While files from the Internet can be useful, some files can potentially harm your computer. If you do not trust the source, do not open or save this file. What's the risk?

Important points to remember about TDM Capture are:

- It is recommended to stop the trace first so the log file gets its header information written before downloading.
- TDM capture occurs regardless of channel call state.
- Capture times
  - Rx only = ~ 48 minutes (data into DMG2000)
  - Rx & Tx = ~ 24 minutes
- Downloading does not require any special software, but viewing requires an audio viewer such as CoolEdit Pro or Adobe Audition (not freeware).

- Downloading the TDM Capture clears the log and starts another.
- Stopping the log and restarting clears the log. **Caution:** This operation is different from the other logs.

9.5 Communicating to the Terminal Interface

Provided you have Admin level privileges, you may communicate with the Media Gateway terminal interface by connecting to the serial interface (DIAGNOSTICS connector on DMG1000 types and COM2 connector on DMG2000 types) located on the rear panel or by establishing a telnet session via the LAN port.

Information in this section includes:

- Connecting to Terminal Interface Via DIAGNOSTICS Connector
- Connecting to Terminal Interface Via LAN Connector

9.5.1 Connecting to Terminal Interface Via DIAGNOSTICS Connector

Perform the following:

1. Connect a serial cable to the serial interface connector on the rear panel of the Media Gateway. For connector pin designation information, refer to the DIAGNOSTICS Connector Pin Designations table (DMG1000) or COM1 and COM2 Connector Pin Designations table (DMG2000) in the *Getting Started Guide*.
2. Using a standard serial interface application (for example, Procomm Plus or HyperTerminal), set the workstation to the following:
   - Baud Rate = 38400 for DMG1000, 115200 for DMG2000
   - Parity = None
   - Data Bits = 8
   - Stop Bits = 1
   - Hardware Flow Control = Off
3. Press the Enter key repeatedly until the following prompt appears:

   PIMG>
4. At the prompt, type **pwd** and press Enter.
5. When prompted, enter the password for the admin user (the default is **IpodAdmin**) and press Enter.

   The Media Gateway will respond with **Admin level accepted**, and then display the prompt. You may now issue terminal commands to the Media Gateway.

9.5.2 Connecting to Terminal Interface Via LAN Connector

Perform the following to initiate a Telnet session at the workstation connected to the Media Gateway via the LAN interface:

1. From the command prompt, type **telnet** followed by a space and the IP address of the Media Gateway. Then press Enter.
2. The **telnet client connected** message will appear. Press Enter. The following prompt will appear:

   PIMG>
3. At the prompt, type **pwd** and then press Enter.
4. When prompted, enter the password for the Admin user (the default is **IpodAdmin**) and press Enter.

   The Media Gateway will respond with **Admin level accepted**, and then display the prompt. You may now issue terminal commands to the Media Gateway.

9.6 Trace Mechanism

The trace mechanism has been designed to provide the maximum flexibility in tracing the various modules of the Media Gateway software. The trace utility is supported by the serial port terminal as well as the telnet terminal, and is available to users at Admin levels and above. Therefore, to enable the trace utility, the user must log on to the terminal using a valid password.

Note: The trace mechanism can be accessed via the Trace Capture interface as described in Section 9.4.2, "Trace Capture", on page 286.

Information in this section includes:

- Trace Format
- Trace Utility
- Trace Commands
- Examples of Trace Commands and Displays

The trace function allows the user to enable or disable different types of traces in different software modules. It also allows the user to enable or disable traces that are related to the telephony ports on the Media Gateway. Therefore, control of telephony port traces is separate from the control of traces of the various software modules. This allows the user to set the trace control of the modules as desired, and then separately set the telephony ports that are needed to be traced.

Traces that are related to a telephony port number are tagged with the 1-based telephony port number to which they are related. For instance:

Telephony port 1 events:

```
002:53:542 [Tel-1    ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:53:782 [Tel-1    ] Event    0:0:48|            250         |
002:53:784 [Tel-1    ] Event    Cpid (250->) (Direct)
```

Telephony port 2 events:

```
002:59:314 [Tel-2     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:59:550 [Tel-2     ] Event    0:0:48|            251        |
002:59:552 [Tel-2     ] Event    Cpid (251->) (Direct)
```

Time-stamping of the trace output can be enabled or disabled by the user.

The telephony switch protocol can be traced, but the output is encoded for security.

9.6.1 Trace Format

The following is a description of the format of a trace display. The example display is broken down into its components.

Example of trace display:

```
003:02:372 [Tel-2     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
```

Components:

003:02:372 - time-stamp in minutes:seconds:milliseconds

[Tel-2] - trace-key and telephony port (no port if non port-related)

Event - trace type

Lamp 60:CallApp0:0 FLASH->OFF

9.6.2 Trace Utility

The trace utility separates the control of software module trace output using trace keys. A single trace key represents a functional software module. The trace utility then uses trace types that can be enabled or disabled for each trace key. The trace keys and trace types are used to enable or disable traces using the trace utility.

9.6.2.1 Trace Key

The trace key defines the specific sub-system within the Media Gateway to trace. For example, if you are only interested in capturing SIP packets (SIP Emulating or SIP Driving mode), you would use the VoIp trace key:

```
trace voip <trace type> on
```

Table 19 describes all trace keys supported by the trace function as well as the trace types available to each trace key.

Table 19. Supported Trace Keys

Trace Keys	Description	Prot	Code	Error	Warn	Init	Alarm	Event	Stat
System	OS-Independent layer, System initialization		●	●	●	●	●		
Tel	Digital Telephony Interface (for protocol, hook, lamp, display, CPID, in phone-emulation and phone-driving etc.)	●(8)	●	●	●	●	●	●(6)	
TelDrv	Low layer ISDN trace coming from the Telesoft driver. Includes the RAW ISDN layer 3 packets. (14)	●(15)	●	●	●	●	●	●	
VoIp	Voice Over IP Software Stacks	●(1)	●(10)			●			
DspCpi	Dsp Media/Call Progress API (media session setup, RTP stats)		●	●	●	●	●	●	●(7)
DspIf	DSP Media Interface (14)								
CadDet	Dsp Call Progress (14)		●	●	●	●		●	
DspRoot	DSP OS-independent layer		●	●	●				
DspVad	DSP Voice Activity Detector		●	●	●			●	
DspDmi	DSP Management Interface		●	●	●				
DspDlmm	DSP Management Interface		●	●	●				
DspNw	Network Interface for DSP (13)		●	●	●				
DspUdp	UDP/IP Protocol Stack handling for DSP (13)		●	●	●				
DspIarp	OS independent ARP interface (13)		●	●	●				
Arp	ARP Interface for DSP (14)			●					
EthMgr	Ethernet Manager for DSP (14)		●	●		●			
Gcc	Generic Call Control API		●	●	●	●			
Gw	Gateway Application State Machine		●	●	●	●	●		
Cfg	Configuration Module		●	●	●	●	●		
Web	Web Interface		●	●	●	●	●		
Si	Serial Interface API (2)	●(9)	●	●	●	●			
Smwi	Serial Interface MWI Device (2)		●	●	●	●			
SiIp	Serial Interface IP Layer (2)		●	●	●	●	●		
Pbn	Packet-Network Interface		●	●	●	●	●		
SipCsm	SIP State Machine		●	●	●	●	●		
Adept	Adept display parsing		●	●	●	●			
CallLog	Call Log Handler							●	
Alarm	Alarm Handler		●	●	●	●			
Web	Web Server		●	●	●				

Table 19. Supported Trace Keys (Continued)

Trace Keys	Description	Prot	Code	Error	Warn	Init	Alarm	Event	Stat
Log	Diagnostics Logging Module (aka Enhanced Diagnostics)		●	●	●				
Phd	Phone-Driving GCC Layer (3)		●	●	●	●	●		
PhnDrv	Phone-Driving State Machine (3)		●	●	●	●	●		
iNimDrv	NIM (telephony network) Driver (3,4)	●(11)	●	●	●	●	●		
Dcif	Direct-Connect Interface (5)		●	●	●	●	●		
NimCtlDc	NIM/DirectConnect Interface (5)		●	●	●	●	●		
PBXTest	PBX Self Test Application (12)		●	●	●	●		●	
LTA	LearnTone Application (12)		●	●	●	●		●	

NOTES:
1) Currently SIP only - enable to trace SIP packets
2) Serial Interface API only active if serial mode enabled
3) Active only if in phone-driving mode
4) Active only on AMI DMG1000
5) Active only in Direct-Connect mode
6) enable to see telephony hookswitch, lamps, voice state, CPID, displays, tone detection (call progress and dtmf)
7) enable to trace RTP packet and error statistics during a media session
8) enable to trace telephony protocol packets (encoded in release version)
9) enable to trace serial telephony interface packets (serial server only)
10) enable to trace SIP Trillium code
11) enable to trace iNim/DMG1000 interface protocol
12) Active only if feature is run
13) DMG1000 only
14) DMG2000 only
15) RAW ISDN layer 3 messages

9.6.2.2 Trace Type

Trace type defines the level of tracing. For example, to capture only errors for the VoIp trace key, the trace type would be set to:

```
trace voip error on
```

The following table describes the trace types that are supported:

Table 20. Supported Trace Types

Trace Types	Trace Description
Prot	Protocol
Code	Software/Code
Error	Errors (1)
Warn	Warnings
Init	Software/Device initialization (1)
Alarm	Alarms (1)
Event	Hookswitch, Lamps, Tones, Displays, Button Presses
Stat	Statistics
(1) System starts with these enabled on all trace keys.	

9.6.3 Trace Commands

The trace commands include the following:

- trace show
- trace default
- trace <trace key> <trace type|all> <on|off>
- trace all <trace type|all> <on|off>
- trace <trace key|all> reset
- trace all off
- trace port <1-based port number|all> <on|off>
- trace time <on|off>
- trace -l

9.6.3.1 trace show

The trace show command displays the trace levels of all trace modules and port-related trace settings. The following is an example of a trace show display:

```
PIMG>trace show
Key          Traces
----------   ------------------------
System       Error Init Debug
Cfg          Error Init Debug
Tel          Error Init Debug
DspIf        Error Init Debug
DspRoot      Error Init Debug
EthMgr       Error Init Debug
Arp          Error Init Debug
Web          Error Init Debug
Alarm        Error Init Debug
```

```
Pbn          Error Init Debug
VoIP         Error Init Debug
Gw           Error Init Debug
Gcc          Error Init Debug
Adept        Error Init Debug
teldrv       Error Init Debug
CallLog      Error Init Debug
Log          Error Init Debug

Channel Trace State ('*' enabled, '-' disabled)
-----------------------------------------------
* * * * * * * *
```

9.6.3.2 trace default

The trace default command sets the default tracing of all trace keys, and enables VoIP and telephony event tracing.

```
PIMG>trace default
```

9.6.3.3 trace <trace key> <trace type|all> <on|off>

This trace command enables or disables trace types on specified trace keys.

Use:

```
trace tel all on
```

to enable all telephony traces

Use:

```
trace tel event on
```

to enable telephony control traces

9.6.3.4 trace all <trace type|all> <on|off>

The trace all command enables/disables trace types on all trace keys.

Use:

```
trace all code off
```

to disable all code traces on all trace keys

Use:

```
trace voip code on
```

to enable code traces only on voip stacks

9.6.3.5 trace <trace key|all> reset

This command resets specified (or all) trace keys to their defaults (from system start).

9.6.3.6 trace all off

This command disables all traces, except **Error** and **Debug**, on all trace keys. **Error** and **Debug** cannot be disabled.

9.6.3.7 trace port <1-based port number|all> <on|off>

This command enables or disables port-related traces.

Use:

```
trace port all off
```

to disable all telephony-port related traces

Use:

```
trace port 1 on
```

to enable only telephony port 1 related traces

9.6.3.8 trace time <on|off>

The trace time command enables or disables time stamps on traces.

9.6.3.9 trace -l

This command displays trace-types.

9.6.4 Examples of Trace Commands and Displays

The following are some examples of various trace commands and the resulting displays:

Example 1

Enable tracing of telephony events, such as button presses, lamps, displays, tones, etc.:

```
PIMG>trace tel event on
Ok
PIMG>exit
Good-Bye.
002:53:542 [Tel-1     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:53:782 [Tel-1     ] Event    0:0:48 |                        250          |
002:53:784 [Tel-1     ] Event    Cpid (250->) (Direct)
```

```
002:59:314 [Tel-2     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:59:550 [Tel-2     ] Event    0:0:48 |                        251          |
002:59:552 [Tel-2     ] Event    Cpid (251->) (Direct)
003:02:300 [Tel-1     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
003:02:372 [Tel-2     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
003:02:490 [Tel-1     ] Event    0:0:48 |          MAY 13  7:44 P             |
003:02:570 [Tel-2     ] Event    0:0:48 |          MAY 13  7:44 P             |
```

Example 2

Enable tracing of SIP messages:

```
PIMG>trace voip prot on
Ok
PIMG>exit
Good-Bye.
[Tel-1      ] Event    Lamp 60:CallApp0:0 OFF->FLASH
[Tel-1      ] Event    0:0:48 |                        250                    |
[Tel-1      ] Event    Cpid (250->) (Direct)
[VoIP       ] Prot     <----v=0!
[VoIP       ] Prot          o=phone 26533 0 IN IP4 10.0.1.110!
[VoIP       ] Prot          s=-!
[VoIP       ] Prot          c=IN IP4 10.0.1.110!
[VoIP       ] Prot          t=0 0!
[VoIP       ] Prot          m=audio 49010 RTP/AVP 0 8 101!
[VoIP       ] Prot          a=rtpmap:0 PCMU/8000/1!
[VoIP       ] Prot          a=rtpmap:8 PCMA/8000/1!
[VoIP       ] Prot          a=rtpmap:101 telephone-event/8000!
[VoIP       ] Prot          a=fmtp:101 0-15!
[VoIP       ] Prot          a=sendrecv!
[VoIP       ] Prot
[VoIP       ] Prot     <----INVITE sip:101@10.0.1.215;Transport=udp SIP/2.0!
[VoIP       ] Prot
From:"250"<sip:port1@10.0.1.110:5060>;user=phone;tag=1A1F32463135364100000ABE!
[VoIP       ] Prot          To:sip:101@10.0.1.215!
[VoIP       ] Prot          Content-Type:application/SDP!
[VoIP       ] Prot          Call-ID:01B2270D9C81400000000006@pbxgw.default.com!
[VoIP       ] Prot          CSeq:1 INVITE!
[VoIP       ] Prot          Allow-Events:refer,message-summary!
[VoIP       ] Prot          Expires:120!
[VoIP       ] Prot          Via:SIP/2.0/UDP 10.0.1.110:5060!
[VoIP       ] Prot          Contact:<sip:port1@10.0.1.110:5060>!
[VoIP       ] Prot          User-Agent:DMG1000 !
[VoIP       ] Prot          Max-Forwards:70!
[VoIP       ] Prot          Supported:100rel,timer!
[VoIP       ] Prot          Content-Length:216!
[VoIP       ] Prot          !
[VoIP       ] Prot          v=0!
[VoIP       ] Prot          o=phone 26533 0 IN IP4 10.0.1.110!
[VoIP       ] Prot          s=-!
[VoIP       ] Prot          c=IN IP4 10.0.1.110!
[VoIP       ] Prot          t=0 0!
[VoIP       ] Prot          m=audio 49010 RTP/AVP 0 8 101!
[VoIP       ] Prot          a=rtpmap:0 PCMU/8000/1!
[VoIP       ] Prot          a=rtpmap:8 PCMA/8000/1!
[VoIP       ] Prot          a=rtpmap:101 telephone-event/8000!
[VoIP       ] Prot          a=fmtp:101 0-15!
[VoIP       ] Prot          a=sendrecv!
```

```
[VoIP      ] Prot
[VoIP      ] Prot     ---->SIP/2.0 100 Trying!
[VoIP      ] Prot          Via:SIP/2.0/UDP 10.0.1.110:5060!
[VoIP      ] Prot          Call-ID:01B2270D9C81400000000006@pbxgw.default.com!
[VoIP      ] Prot          CSeq:1 INVITE!
[VoIP      ] Prot
From:"250"<sip:port1@10.0.1.110:5060>;user=phone;tag=1A1F32463135364100000ABE!
[VoIP      ] Prot          To:sip:101@10.0.1.215;tag=3246313536412A29002455E5!
[VoIP      ] Prot          Contact:<sip:101@10.0.1.215:5060>!
[VoIP      ] Prot          Server:DMG1000 1.2!
[VoIP      ] Prot          Supported:100rel,timer!
[VoIP      ] Prot          Content-Length:0!
[VoIP      ] Prot          !
[VoIP      ] Prot
[VoIP      ] Prot     ---->SIP/2.0 180 Ringing!
[VoIP      ] Prot          Via:SIP/2.0/UDP 10.0.1.110:5060!
[VoIP      ] Prot          Call-ID:01B2270D9C81400000000006@pbxgw.default.com!
[VoIP      ] Prot          CSeq:1 INVITE!
[VoIP      ] Prot
From:"250"<sip:port1@10.0.1.110:5060>;user=phone;tag=1A1F32463135364100000ABE!
[VoIP      ] Prot          To:sip:101@10.0.1.215;tag=3246313536412A29002455E5!
[VoIP      ] Prot          Contact:<sip:101@10.0.1.215:5060>!
[VoIP      ] Prot          Server:DMG1000 1.2!
[VoIP      ] Prot          Supported:100rel,timer!
[VoIP      ] Prot          Content-Length:0!
[VoIP      ] Prot          !
[VoIP      ] Prot
```

Example 3

Enable trace of RTP statistics:

```
PIMG>trace dspcpi stat on
Ok
PIMG>exit
Good-Bye.
[DspCpi-1  ] Stat     Receive/Transmit Stats:
[DspCpi-1  ] Stat         Rx Voice Packets:             0
[DspCpi-1  ] Stat         Tx Voice Packets:             2
[DspCpi-1  ] Stat         Tx Silence Suppressed Frames: 0
[DspCpi-1  ] Stat         Rx Min Jitter:                -1 (ms)
[DspCpi-1  ] Stat         Rx Max Jitter                 0 (ms)
[DspCpi-1  ] Stat         Rx RTP Avg Jitter:            0 (pcm samples)
[DspCpi-1  ] Stat         Tx Grant Sync Dropped Frames: 0
[DspCpi-1  ] Stat         Tx Octets:                    160
[DspCpi-1  ] Stat         Rx Octets:                    0
[DspCpi-1  ] Stat         AAL2 Coding Profile Changes:  0
[DspCpi-1  ] Stat         DTMF Tx Packets:              0
[DspCpi-1  ] Stat         DTMF Rx Packets:              0
[DspCpi-1  ] Stat         SID Rx Packets:               0
[DspCpi-1  ] Stat         SID Tx Packets:               0
[DspCpi-1  ] Stat         Tx Last Timestamp:            10313
[DspCpi-1  ] Stat         Tx Extended Seq Number:       0
[DspCpi-1  ] Stat         Tx Last Seq Number(AAL2-UUI): 1
[DspCpi-1  ] Stat         Tx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Timestamp:            0
[DspCpi-1  ] Stat         Rx Last Ssrc:                 0
```

```
[DspCpi-1  ] Stat         Rx Extended Seq Number:       0
[DspCpi-1  ] Stat         Rx Last Seq Number(AAL2-UUI): 0
[DspCpi-1  ] Stat         Pkt Lost by Network:          0
[DspCpi-1  ] Stat         Rx P2P Packets to Hosts:      0
[DspCpi-1  ] Stat         Rx P2P Filtered Packets:      0
[DspCpi-1  ] Stat         P2P Squelched Voice Packets:  0
[DspCpi-1  ] Stat         Rx Net Packets:               0
[DspCpi-1  ] Stat         Tx Net Packets:               2
[DspCpi-1  ] Stat     Error Stats:
[DspCpi-1  ] Stat         invalid_header_count:         0
[DspCpi-1  ] Stat         to_micro_overflow_count:      0
[DspCpi-1  ] Stat         lost_enh_packet_count:        0
[DspCpi-1  ] Stat         no_core_packet_count:         0
[DspCpi-1  ] Stat         pkt_lost_by_network:          0
[DspCpi-1  ] Stat         rc4key_update_lost_pkt_count: 0
[DspCpi-1  ] Stat         invalid_mac_header_count:     0
[DspCpi-1  ] Stat         invalid ssrc count:           0
[DspCpi-1  ] Stat         invalid payload count:        0
[DspCpi-1  ] Stat     Voice Playout Stats:
[DspCpi-1  ] Stat         avg_playout_delay:            20
[DspCpi-1  ] Stat         lost_packet_count:            0
[DspCpi-1  ] Stat         replay_packet_count:          0
[DspCpi-1  ] Stat         idle_packet_count:            0
[DspCpi-1  ] Stat         dropped_packet_count:         0
[DspCpi-1  ] Stat         rx_packet_count:              0
[DspCpi-1  ] Stat         avg_frame_jitter:             2 (ms)
[DspCpi-1  ] Stat         adpt_po_buf_delay_inc:        0
[DspCpi-1  ] Stat         adpt_po_buf_delay_dec:        0
[DspCpi-1  ] Stat         po_buf_underflow_cnt:         0
[DspCpi-1  ] Stat         cell_starve_evt_cnt:          0
[DspCpi-1  ] Stat     Receive/Transmit Stats:
[DspCpi-1  ] Stat         Rx Voice Packets:             151
[DspCpi-1  ] Stat         Tx Voice Packets:             506
[DspCpi-1  ] Stat         Tx Silence Suppressed Frames: 0
[DspCpi-1  ] Stat         Rx Min Jitter:                30 (ms)
[DspCpi-1  ] Stat         Rx Max Jitter                 30 (ms)
[DspCpi-1  ] Stat         Rx RTP Avg Jitter:            0 (pcm samples)
[DspCpi-1  ] Stat         Tx Grant Sync Dropped Frames: 0
[DspCpi-1  ] Stat         Tx Octets:                    40480
[DspCpi-1  ] Stat         Rx Octets:                    36480
[DspCpi-1  ] Stat         AAL2 Coding Profile Changes:  0
[DspCpi-1  ] Stat         DTMF Tx Packets:              0
[DspCpi-1  ] Stat         DTMF Rx Packets:              0
[DspCpi-1  ] Stat         SID Rx Packets:               0
[DspCpi-1  ] Stat         SID Tx Packets:               0
[DspCpi-1  ] Stat         Tx Last Timestamp:            50633
[DspCpi-1  ] Stat         Tx Extended Seq Number:       0
[DspCpi-1  ] Stat         Tx Last Seq Number(AAL2-UUI): 505
[DspCpi-1  ] Stat         Tx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Timestamp:            67547
[DspCpi-1  ] Stat         Rx Last Ssrc:                 1368282060
[DspCpi-1  ] Stat         Rx Extended Seq Number:       0
[DspCpi-1  ] Stat         Rx Last Seq Number(AAL2-UUI): 151
[DspCpi-1  ] Stat         Pkt Lost by Network:          0
[DspCpi-1  ] Stat         Rx P2P Packets to Hosts:      0
[DspCpi-1  ] Stat         Rx P2P Filtered Packets:      0
[DspCpi-1  ] Stat         P2P Squelched Voice Packets:  0
[DspCpi-1  ] Stat         Rx Net Packets:               152
[DspCpi-1  ] Stat         Tx Net Packets:               506
```

```
[DspCpi-1  ] Stat     Error Stats:
[DspCpi-1  ] Stat         invalid_header_count:         1
[DspCpi-1  ] Stat         to_micro_overflow_count:      0
[DspCpi-1  ] Stat         lost_enh_packet_count:        0
[DspCpi-1  ] Stat         no_core_packet_count:         0
[DspCpi-1  ] Stat         pkt_lost_by_network:          0
[DspCpi-1  ] Stat         rc4key_update_lost_pkt_count: 0
[DspCpi-1  ] Stat         invalid_mac_header_count:     0
[DspCpi-1  ] Stat         invalid ssrc count:           1
[DspCpi-1  ] Stat         invalid payload count:        0
[DspCpi-1  ] Stat     Voice Playout Stats:
[DspCpi-1  ] Stat         avg_playout_delay:            27
[DspCpi-1  ] Stat         lost_packet_count:            0
[DspCpi-1  ] Stat         replay_packet_count:          0
[DspCpi-1  ] Stat         idle_packet_count:            0
[DspCpi-1  ] Stat         dropped_packet_count:         0
[DspCpi-1  ] Stat         rx_packet_count:              453
[DspCpi-1  ] Stat         avg_frame_jitter:             1 (ms)
[DspCpi-1  ] Stat         adpt_po_buf_delay_inc:        0
[DspCpi-1  ] Stat         adpt_po_buf_delay_dec:        0
[DspCpi-1  ] Stat         po_buf_underflow_cnt:         0
```

Example 4

Enable trace of telephony protocol:

```
PIMG>trace tel prot on
Ok
PIMG>exit
Good-Bye.
[Tel-1     ] Prot
PIMGXX00152BB986F7A64E5028A7A1E69B2C7FA0D56681CB1D9F92D287C80C3F5D96593
[Tel-1     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
[Tel-1     ] Prot
PIMGXX01171850BBCFB4660E32E3C0FC67C53DF45CB602FC847DCF07E52F678A1E8153EA9
[Tel-1     ] Prot     PIMGXX010744A
[Tel-1     ] Prot
PIMGXX021790EDBC5B8E6F74C62762B8BE5CD5A11DDE49E8BD7AACA72910DAE88013BB77D
[Tel-1     ] Prot     PIMGXX020610FA12D05
[Tel-1     ] Prot
PIMGXX0310E4CD136BB199A9C4596613EFB98084BB5EF810030EBEE7A1A8CD5CBA1A81831
[Tel-1     ] Prot     PIMGXX0307F2B
[Tel-1     ] Prot
PIMGXX041E8AA914C95D9F88E5210FA3395984BD28D205B67AC46B4281E0AE991BE6606
[Tel-1     ] Event    0:0:48 |                        250                    |
[Tel-1     ] Event    Cpid (250->) (Direct)
[Tel-1     ] Prot
PIMGXX0511126EB93134DD19188F9BE6A62682B82F6D4083E8DBBB1C5D7ABAF688009BC
[Tel-1     ] Prot
PIMGXX0612911CB3EE87834C804848D1611B468DAA9496CEF645A59429ADC64505C5BE8FB
[Tel-1     ] Prot     PIMGXX0605CCB
[Tel-1     ] Prot
PIMGXX071EF122ABB171D6D0CD98074220E19E8901C84B68AE601B1372923FE2255E19E
[Tel-1     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
[Tel-1     ] Prot
PIMGXX081DF265B1B82C6B322456F4E979ABF50492CDE81A1CC65EE64FDD407E29D08A5
[Tel-1     ] Event    0:0:48 |          MAY 13  7:46 P                      |
```

9.7 Diagnostic Commands

This section describes a number of diagnostic commands and includes:

- Devstat Command
- Restart Command
- Ping Command
- Ver Command
- Alarm List Command

9.7.1 Devstat Command

The Devstat command displays the current high-level settings of the unit and the unit's main status and telephony port status. The telephony port state can be 'In Service', 'Out of Service', or 'Active'. 'Active' means that the port is off-hook. 'In Service' means that the port is available for use, but is currently not in use. 'Out of Service' means that the port is disconnected.

```
PIMG>devstat
Description       Value
=========================================================================
sysStartType      0x0
MAC               00-a0-e6-02-01-19
IP                10.0.1.110
Subnet Mask       255.255.255.0
Gateway           10.0.1.1
Host              10.0.1.63
Status            OK

Port 1            Active
Port 2            In Service
Port 3            In Service
Port 4            In Service
Port 5            In Service
Port 6            In Service
Port 7            In Service
Port 8            In Service
```

9.7.2 Restart Command

The Restart command causes the unit to warm-boot.

```
restart
```

9.7.3 Ping Command

The Ping command generates ICMP requests to the specified IP address. This is useful in determining connection status to IP endpoints, gateways, or routers.

```
PIMG>ping 10.0.1.1
10.0.1.1 is alive
PIMG>ping 10.0.1.1 5
PING 10.0.1.1: 56 data bytes
64 bytes from 10.0.1.1: icmp_seq=0. time=2. ms
64 bytes from 10.0.1.1: icmp_seq=1. time=0. ms
64 bytes from 10.0.1.1: icmp_seq=2. time=0. ms
64 bytes from 10.0.1.1: icmp_seq=3. time=0. ms
64 bytes from 10.0.1.1: icmp_seq=4. time=0. ms
----10.0.1.1 PING Statistics----
5 packets transmitted, 5 packets received, 0% packet loss
round-trip (ms) min/avg/max = 0/0/2
PIMG>ping 10.10.10.10
no answer from 10.10.10.10
```

9.7.4 Ver Command

The Ver command outputs the versions of all hardware and firmware of the unit.

```
PIMG>ver
Description       Version
=========================================================================
DSP               |9.1| |THU AUG 29 15:22:54 2002|
Flash App         |lab_D| |THU NOV 21 16:20:47 2002|
Flash DSP         |9.1| |THU AUG 29 15:22:54 2002|
Boot              |dhcp| |WED NOV 20 12:48:16 2002|
CPLD              0x21
iNIM App          6.6
iNIM FPGA         179
iNIM Boot         5.0
Base HW           2
```

9.7.5 Alarm List Command

The Alarm List command outputs the current list of generated system alarms. The timestamp of each alarm is the time since unit start in hours:minutes:seconds.

```
PIMG>alarms
000:00:000        System            MIB-II Trap: Cold start (0)
000:00:000        System            MIB-II Trap: Cold start (0)
000:12:000        Telephony         snw (1) - In Service
000:12:000        Telephony         snw (1) - In Service
000:12:000        Telephony         snw (2) - In Service
000:13:000        Telephony         snw (4) - In Service
000:13:000        Telephony         snw (5) - In Service
000:13:000        Telephony         snw (6) - In Service
000:13:000        Telephony         snw (7) - In Service
000:13:000        Telephony         snw (8) - In Service
000:15:000        Telephony         snw (3) - In Service
```

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